

SECTION 7

OPERATING NOTES AND HINTS

This section is not meant to be comprehensive. Instead, it focuses on a few areas which we feel require special attention, or where a better understanding of the function can lead to far more utility or better sound quality from the PM2800M.

7.1 CONSOLE GAIN STRUCTURE

In the GAIN STRUCTURE AND LEVELS section of this manual, we discuss some general considerations regarding levels and system setup. What of the proper gain structure within the PM2800M? How can the many faders and other level controls that affect a given signal all be adjusted for the optimum results? These are important questions to ponder, and we hope you will take some time to study the possibilities.

7.1.1 What is the Proper Gain Structure?

Let's begin with the XLB channel input to the console. According to the INPUT CHARACTERISTICS chart in the SPECIFICATIONS section, the nominal input level ranges from -60 dBu (0.75 mV) to +4 dBu (1.23 V). These are the levels that will supply the ideal signal level throughout the module with the PAD set to 0 dB or -40 dB, the input GAIN control as required, AND fader set to its nominal position. Actually, a wider range of levels can be accommodated if the GAIN control and fader are further adjusted; from -80 dBu (0.075 mV) minimum to +24 dBu (12.3V) maximum.

What is the correct gain structure? Simply stated, it is the level at which there remains adequate headroom so that peaks can be accommodated without clipping, while at the same time there is sufficient "distance" above the noise floor that noise does not become objectionable. If a signal is too high in level (too "hot") at a given point in the console, then peaks or, in the extreme, the entire signal, will be subject to distortion. If the signal is too low in level, there may be considerably more headroom and less risk of distortion, but the noise will be that much more noticeable, and quiet passages may be masked entirely by residual noise. The "ideal" level, then, where headroom and noise tradeoffs are optimum, is also known as the nominal level. There is no single value for the correct nominal level; it varies throughout the console. This is what the middle graph line in the GAIN STRUCTURE chart in Figure 3-22 depicts. The top graph line indicates the clipping point. The

distance between these two lines, at any point along the horizontal signal flow scale, depicts the available headroom. It is important that wide headroom be available throughout a console, not just at the input and output; otherwise multiple signals applied to the busses may add together such that the mixed level approaches clipping, even though the individual feeds to the mix are within their acceptable nominal range. Sometimes a mix or master fader can be adjusted to correct this condition, other times it cannot because the distortion is occurring in an amplifier ahead of the fader, and the only cure is to lower the signal levels applied to the bus. How can one know the best course of action when distortion, or excess noise, is encountered?

7.1.2 What Affects Gain Structure?

First, understand that signal levels can be increased by either increasing amplifier gain (including EQ boost), reducing the amount of attenuation, or adding multiple signals together. Similarly, signal levels can be reduced by either decreasing amplifier gain (which includes increasing the amount of EQ cut), increasing the amount of attenuation (including filter roll-off), or splitting the signal to feed two or more circuits. With this in mind, it becomes clear that the mere act of feeding the "correct" nominal level signal into a console is no guarantee that it will remain at an acceptable level throughout the console.

7.1.3 Establishing the Correct Input Channel Settings

In the case of the PM2800M, the pair of SIGNAL and CLIP LEDs adjacent to the input channel PAD and GAIN controls make it relatively simple to obtain the correct gain structure at the input stage. Begin with the PAD set at maximum attenuation (-40 dB), the GAIN control centered, and apply the typical input signal to the channel input. If the green SIGNAL LED is not on, adjust the attenuation PAD switch to lower attenuation values until the SIGNAL LED turns on. Then try even lower attenuation PAD settings to see if you can make the red CLIP LED flash on regularly or almost continuously; if this occurs, move the PAD back one notch to a higher setting where the CLIP LED may flash only occasionally or not at all. If, at the minimum attenuation (0 dB) PAD setting the CLIP LED flashes occasionally or not at all, then leave the PAD in this position. Next, adjust the GAIN control as required so that the red

CLIP LED flashes on only occasionally, during the louder program peaks. This establishes the correct channel sensitivity for the initial setup (you may wish to alter these values during an actual program mix, as explained in subsequent paragraphs). A more precise input sensitivity setting can be obtained by placing the console in SOLO mode [58] (*do this prior to the actual performance*), engaging the CUE meter switches [74], and pressing the channel CUE/SOLO button [19].

NOTE: It is a good idea to set the Mix Master Faders, the Stereo Master, the Matrix Master controls and all Aux Send Faders at a very low level during the initial stages of setup. This will prevent uncomfortable or even dangerously loud signals from reaching the outputs while preliminary mix setup is established.

Given the correct GAIN and PAD settings, adjust the channel Fader to its nominal (0 dB) setting. This setting provides the best range of control, with some boost available if the signal must be raised in the mix, and plenty of resolution for fading the signal down in the mix.

Now the channel HP Filter and EQ can be set as desired. If a particular EQ setting causes the EQ CLIP LED to flash on more than occasionally, then the boost applied is raising the signal level too high. The solution is to either reduce the EQ boost setting in one or more bands, or leave the EQ where you have it for the proper signal contour, and instead reduce the signal level going into the equalizer. You must do this by adjusting the GAIN control (and/or PAD); the Fader does not affect signal going into the EQ. Lower the GAIN only enough so that the EQ CLIP LED does not flash on excessively.

The signal now may be assigned to any of the eight primary mixing busses or the four auxiliary mixing busses (we describe the stereo bus in the next subsection). Assuming the aux sends are in the factory-shipped pre-fader mode, then the signal level applied to those aux busses will remain constant regardless of adjustments to the channel Fader, depending instead only on the AUX control setting. If the aux send jumper has been moved to the post-fader mode, the send level will be determined by both the channel AUX control and the channel Fader.

This same procedure should now be followed for each input channel. Once this is done, the bus levels can be examined. Monitor the mix which will be used for the solo (lead) vocalist or other lead musician (use headphones plugged into the PHONES jack [69] and engage the mix bus' CUE switch [36]). Adjust the mix

control [8 and/or 6, 7] corresponding to that bus on each input channel which contributes to the mix. Initially, set these mix controls at approximately #7 on their scale (0 – 10), then readjust them as necessary to create a rough mix of all input channels which feed this mixed output. Bring down the input channel mix controls [8 and/or 6, 7] for those sources which are too prominent in the mix; avoid adjusting the input Faders [18] at this point. Once this rough mix is established, raise the bus' Master Fader (i.e., Mix 1 – 8 or Aux Send 1 – 4) toward the nominal position (0 dB on the scale). Look at the appropriate VU meter (i.e., for MIX OUT or AUX OUT). If the signal level on this output bus becomes too hot (red LED flashing on more than occasionally or VU meter pegged at the top of the scale), do not back off the Master Fader. Instead, readjust all the input channel mix controls which feed this output mix by an equal amount. This will leave the output Master Fader at the desired nominal position, preserve the desired balance between input channels, and keep the bus level from being too hot. Finally, release the output mix CUE switch.

7.1.4 Establishing a Stereo Mix

This procedure is similar to that for the previously described output mixes. The STEREO mix controls on each input [9] are factory wired to derive the signal post-fader, so be sure the channel fader is set to nominal (zero dB) position. [If the stereo mix has been internally jumpered to pre-fader mode, then the fader position does not matter here.] To monitor the Stereo output, press its CUE switch [57], and then bring up the input's STEREO LEVEL control [9, center knob] and adjust the PAN control [9, outer knob] as need be. Do the same for each input channel you wish to assign to the stereo mix. If you're not sure about the stereo position of a given input source once several channels are assigned to the stereo mix, you can temporarily place the console in the SOLO mode [58], then press the uncertain input's CUE/SOLO switch [19], and you will hear only that source so you can more accurately adjust its position in the stereo field. With the various signals applied to the stereo mix, bring up the STEREO Master Faders [56] to nominal position and check the bus levels on the STEREO L and R VU meters; if they are too high, you can lower all input channel STEREO LEVEL controls [9] by the same amount in order to preserve the mix balance. If the overall left or right level is imbalanced, you can offset one of the two STEREO Master Faders relative to the other.

7.1.5 Establishing the Correct Aux Send Master Fader Settings

It is now appropriate to adjust the AUX Send Master Faders [29]. Assuming the internal Aux Send jumpers have not been changed to pre-fader, leave the input channel Faders [18] set at nominal position (0 on the fader scale), and adjust only the AUX controls [6, 7] on all the inputs that feed a given aux bus to obtain the optimum mix. Monitor that bus mix with the corresponding Aux Send CUE switch [30], and then bring up the associated AUX Send Master Fader [29] to nominal level (0 dB). If the AUX OUT VU meter indicates the bus level is too high, back off on all the correspondingly numbered input channel AUX controls, not the AUX Send Master Fader. Release this Aux Send CUE switch [30], and repeat the same procedure for each of the remaining AUX Sends.

7.1.6 Establishing the Correct Mix Matrix Settings

Since the matrix is fed from the mix busses and stereo bus, its gains should be adjusted only after the Mix Master and Stereo Master Faders have been calibrated. (It makes little difference to the sequence of events whether the MIX-TO-MATRIX and ST-TO-MATRIX assignments are pre or post Master Fader (these pre/post Master Fader assignments are changeable via internal jumpers); the Mix and Stereo bus calibrations still must be done first to establish the proper levels on these busses ahead of the Master Faders.

Here, a similar approach can be used, monitoring the matrix outputs one at a time with the Matrix CUE switches [40], adjusting any individual matrix controls [37, 38] you wish to include in that matrix mix first to the nominal setting (Nº 7 on the scale), then reducing the setting of some of these controls to obtain the desired mix, and finally bringing up the MATRIX MASTER control [39] to nominal position (Nº 7) and, if necessary, reducing the contributing matrix mix controls by an equal amount to avoid too-high matrix output bus levels.

7.1.7 Establishing the Correct Aux Return Settings

With the aux sends calibrated, any external signal processors (effects units such as reverbs, delay lines, phasers, etc.) which are fed from the aux system can be adjusted for optimum input and output levels. Assuming the auxiliary processors are brought into one or more of the PM2800M AUX RTN inputs, those returns are ready to be calibrated. The CUE switches for each Aux Return [26, 27] are of little value here because they derive signal ahead of the Aux Return LEVEL control. Instead, monitor any bus(es) to which the AUX RTN is assigned, and set that aux return input's LEVEL control [25] for the desired mix level. If the LEVEL control is at the maximum (Nº 10), and the signal applied to the AUX RTN input is too low in level, then the output level of the auxiliary processor should be increased. If the LEVEL control is below 1/3-rotation (about Nº 3), then the output of the auxiliary processor should be attenuated somewhat so that the PM2800M Aux Return LEVEL [25] can be raised closer to nominal (pointer mark). Once all AUX RTN inputs have been so calibrated, it is possible that the additional signal strength they have contributed to any busses to which they are assigned may have raised the overall bus level too high. Again check the VU meters on affected MIX, STEREO or AUX busses. In this case, the bus Master Faders may be used for minor "touch up" level adjustments. If the level is much too high on a given bus, do not pull down its Master Fader more than a few dB; instead, lower the input Level controls (or Faders) for all channels which contribute to that "hot" bus.

7.1.8 Channel Muting and Gain Structure

As pointed out earlier, adding inputs to a mix will increase mix bus levels. If optimum mix levels are established while some input channels are muted, and those channels are later turned on (either with the channel ON/off switch or by using the channel MUTE and MASTER MUTE switches), then the bus levels may increase unacceptably. Should this occur, all input channels' levels applied to the offending bus or busses may have to be reduced. Similarly, if some Mixes are added to the Mix Matrix after those gains have been calibrated, then the Matrix levels may increase unacceptably, requiring either a reduction in all Mix Master Faders [35, 56] (for post-master matrix feeds), a reduction in all input mix controls feeding the busses [8, 9] (for pre-master matrix feeds) and/or a reduction in the settings of the Matrix Mix controls [37, 38]. Minor adjustments of the MATRIX MASTER controls [39] should be the last item, if necessary.

7.2 FURTHER HINTS & CONCEPTUAL NOTES

7.2.1 Understanding and Using the Mix Matrix

The PM2800M Mix Matrix consists of 10 smaller mix level controls [37, 38] and one larger MATRIX MASTER control [39] on each of four modules (the "Mix A" and "Mix B" modules). These 44 controls can be thought of as a small mixer within the larger console. In general, the matrix is used to create different output mixes from the same set of mixing busses. The matrix is considerably more convenient and less costly than actually using an external line mixer, and in the case of the PM2800M, it is more flexible as well.

Let's "walk through" the PM2800M mix matrix. Each matrix "channel" (a vertical row of controls) is identical. All eight Primary Mix busses plus the Stereo Mix bus are mixed to a mono signal using the individual matrix mix level controls. The overall level of the mix of these 10 sources can be adjusted with the MATRIX MASTER control.

If you examine the block diagram of the matrix provided in Figure 7-1, you will see that the level adjustments made in one channel of the matrix affect only that matrix output. They do not affect levels in any other matrix channel, nor do they affect any other console outputs. On the other hand, assuming the signals are fed to the matrix after the Mix Master Faders [35] and the STEREO Master Fader [56] (which is how the PM2800M is supplied from the factory), then adjustments of the Mix and Stereo bus output levels will affect the levels applied to the matrix.

NOTE: The signal fed from each Mix bus and from the Stereo bus to the matrix is factory wired so that it is derived after the Master Faders. A jumper wire associated with each Master Fader may be reset so that the feed to the matrix is derived ahead of the Master Fader (see Section 6). In that case, the Mix or Stereo Fader setting would not affect the matrix levels.

The four matrix channels can be used to create four different 10:1 mono mixes, or they may be used to create two different 10:2 stereo mixes, or a combination of mono and stereo mixes. These multiple mixes can be used for a variety of purposes, depending on the application. See the APPLICATIONS section of this manual for specific examples regarding the matrix.

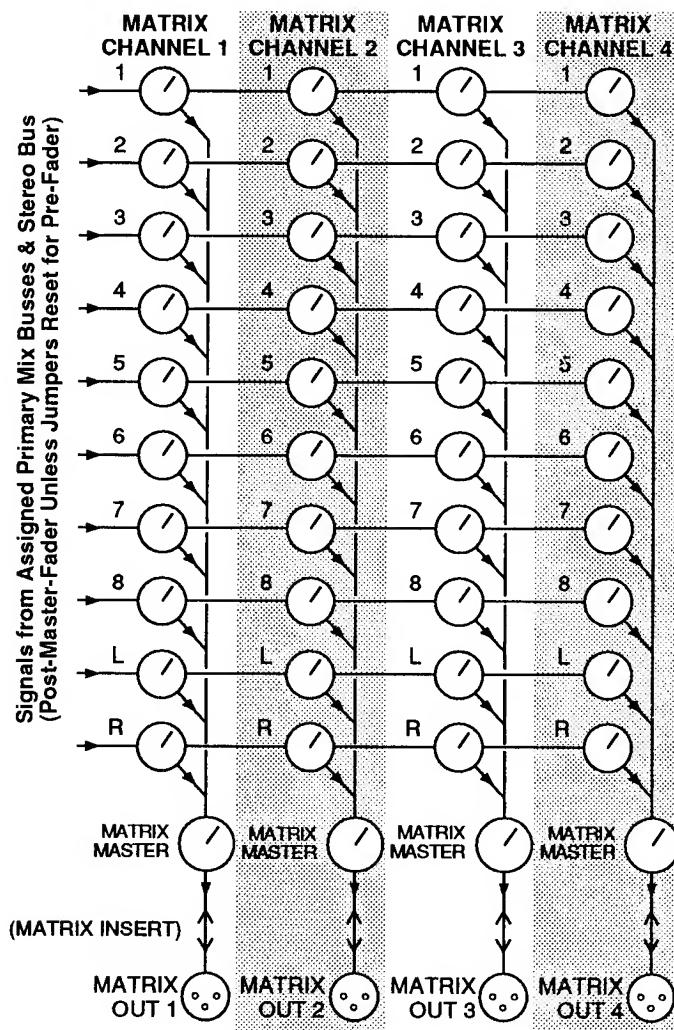


Figure 7-1. Block Diagram of PM2800M Mix Matrix.

7.2.2 Understanding and Use of the Master Mute Function.

Each input channel is provided with eight MUTE Assign switches [17]. When one of these switches is engaged on a given input channel, that channel becomes subject to control by the correspondingly numbered MUTE MASTER switch [68]. Specifically, when the MUTE MASTER switch is engaged, then the assigned input channel(s) turn Off (assuming they had been turned On in the first place). What this means is that any assortment of input channels can be preset to turn off when one or more of the MUTE MASTER switches is engaged (or to turn on when the MUTE MASTER switch is released). This is useful in just about every conceivable application.

In a concert, an entire group of inputs can be muted when the instrument players and/or vocalists are not using them. The input channel faders and other mix controls can all be left at their previously established settings, and only one MUTE MASTER switch need be engaged to keep these mic or line level sources from contributing to the console output. Then,

at the precisely required moment, that group of channels can be brought into the mix "on cue" by releasing the appropriate MUTE MASTER switch(es).

For a theatrical presentation, different scenes can be unmuted as required, keeping the number of open mics at a minimum, which reduces the tendency for feedback with distant mics. For a varied musical presentation, a group of inputs which are primarily used for solo performances can be kept muted until the moment they are needed, thus minimizing noise. For a TV production, on-stage mics can be kept muted until the moment the participant is called upon to use the mic, thus reducing noise, the "hollow" sound from those open mics, and removing the extra stress on the participants. These are but a few of the ways that the PM2800M's ability to mute overlapping groups of input channels can be used to advantage.

NOTE: While a similar function could be achieved by using the Mix ON/off switches, the functions are really different. Consider that the MUTE MASTER switch kills all of the outputs

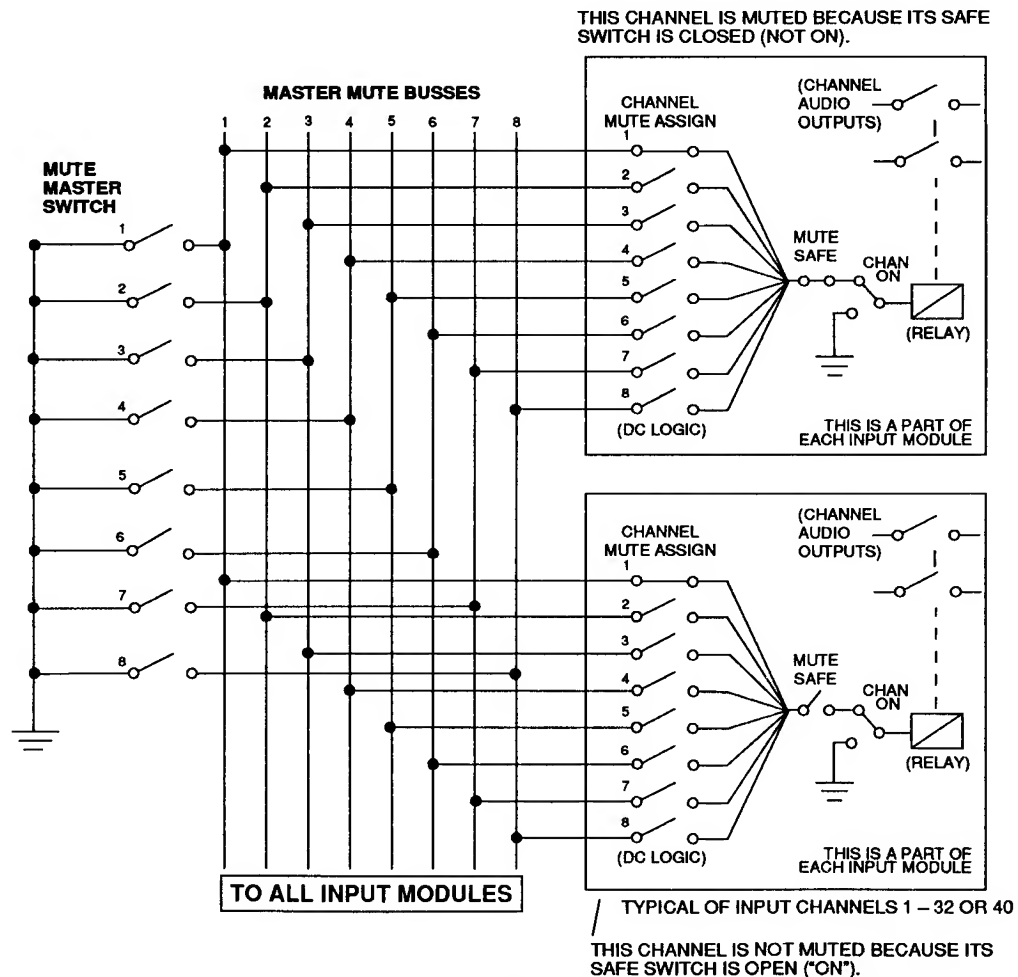


Figure 7-2. Block Diagram of PM2800M Master Mute System.

from the input channels, including the aux sends, whereas each Mix ON/off switch kills only one mix output. Also, consider that some input channels feeding a given mix can be killed with one MUTE MASTER, while other input channels may continue feeding that mix output. Thus, the mute function is distinctly different than the Mix output ON/off switches.

Things can become more complex when an input channel is assigned to more than one MUTE MASTER switch. In this case, the mere act of releasing one MUTE MASTER may not turn on the channel — if the channel is still being muted by so much as one other assigned MUTE MASTER. Should the need arise to turn on a particular input channel without unmuting other channels, and you don't want to disturb the previously assigned MUTE switches, you can override the entire muting system by engaging that channel's MUTE SAFE switch [16]. MUTE SAFE, in effect, blocks any of the channel's MUTE ASSIGN switches [17] which assures that the channel will remain on so long as its ON/off switch [15] is engaged.

7.2.3 Stereo Mixing Considerations

The console's bus assignment is very flexible. The input channel PAN pot will locate the channel output in Left and Right stereo mixing busses.

There are instances when more than one stereo mix will be required. In such applications, pairs of mixing busses can be used to create additional stereo mixes. For example, suppose the normal stereo mix bus is used for sidefill monitors, but the keyboard player also needs a stereo monitor mix. In this case, it may be convenient to use two of the primary mix busses as though they were Left and Right stereo busses; let's assume bus 7 is "keyboard left" and bus 8 is "keyboard right." Any input channels you wish to assign to the keyboard player's stereo monitors can then be assigned with mix controls 7 and 8 [8]; for full-left position, turn up the channel's mix 7 control all the way (Nº 10 on the scale), and leave the mix 8 control turned down (zero on the scale). Conversely, for full-right position, turn up mix 8 and leave mix 7 down. For center position, turn both controls up to about Nº 8 or so (not all the way up). The Mix 7 and Mix 8 Master faders [35] now become a stereo master output fader for this mix. (If the keyboard player has more than one set of keyboards, this same stereo mix can be routed to separately-adjustable outputs via the mix matrix).

Lets say the drums are being mixed in stereo, and must be compressed as a stereo mix. One does not want the drum compression to affect the other pro-

gram material, so compressing the stereo output is not the answer. Therefore, the drum input channels can be assigned to a pair of adjacent odd and even numbered mix busses, creating a stereo mix. The INSERT IN/OUT jacks of those two mix busses are then patched to a stereo compressor/limiter, such as the Yamaha GC2020B, which affects only the stereo drum mix. The two mixes are then mixed together with the STEREO mix, which is done using the matrix (the MIX-TO-MATRIX switches [32] and STEREO-TO-MATRIX switch [53] must be engaged). Two of the matrix channels can then be used to combine the primary STEREO mix with the auxiliary mix created on the pair of mix busses. Using this approach, up to 4 mix-bus-generated stereo mixes can be processed independently of each other, then mixed with the stereo bus into a single stereo program. Alternately, the separate stereo programs can be used for completely different purposes and never mixed together (one for a recording feed, one for the house, one for keyboards, one for sidefill, etc.)

7.3 INTERFACE WITH POPULAR INTERCOM SYSTEMS

In the Introduction to this manual, and in the "Brief Operating Instructions" in Section 2, we mentioned that the PM2800M can be tied to popular intercom systems so that the console operator need wear just one headset for cue and intercom. Intercom systems manufactured by Clear-Com and by RTS Systems utilize different types of connections, and each of these manufacturers is sufficiently popular to be considered a "standard." The following instructions are provided to facilitate interface with these systems, and with other systems using the same type of connections.

To begin, we will review standard intercom terminology. In these terms, the console utilizes a "4-wire" communication circuit. That is, a signal *comes out* of the console to the intercom system on one pair of wires — the "Transmit" pair— and *goes into* the console on a second pair of wires— the "Receive" pair. The shields are not significant for the purpose of this definition.

In Yamaha consoles such as the PM1800, PM2800M, or PM3000, the "Transmit" circuit (*to* the intercom system) is provided by the Talkback Output (TB OUT) while the "Receive" circuit (*from* the intercom system) is provided by the COMM IN input.

On the other hand, most popular intercom systems, such as those from Clear-Com and RTS, are referred to as "2-wire" communications circuits. This means

that, unlike typical audio circuits, the 2-way intercom conversation simultaneously travels in two directions at once on a single pair of wires. This method of operation is also known as “Full Duplex,” and is just like that used in telephone systems.

The “4-wire” console circuit can be connected to the intercom system by one of two methods:

1. The TB OUT (“Transmit”) and the COMM IN (“Receive”) can be connected to the intercom *line* through the appropriate isolation networks. This method results in the intercom operator hearing his/her own voice on the cue circuit at a very high level if he/she is simultaneously listening and talking into the intercom system. (There is no “Anti-Sidetone” function to reduce the voice level coming back to the operator, as there is in the typical intercom station.)
2. The TB OUT (“Transmit”) and the COMM IN (“Receive”) can be connected to the intercom *line* through a “4-wire to 2-wire” interface/converter. (The “4-wire” represents the console, “2-wire” the intercom.) This interface or converter may be obtained from the intercom manufacturer, or it can be built using passive, level-matching components so that the console can be connected to the headset connector of an intercom station. This method of interface provides the optimum anti-sidetone function, which allows the console to operationally emulate a typical intercom station.

7.3.1 RTS Intercom Systems

RTS utilizes a 3-wire interconnect between their standard intercom stations. They also use standard XLR-3 connectors in most cases so that ordinary microphone cables may be employed. (In some cases, XLR-4 connectors are used). However, DC power is superimposed on the audio signal within the RTS intercom system, and there are actually two audio channels carried in the 3-wire cable, so one cannot

necessarily simply plug a standard mic cable between the PM2800M and the RTS intercom (except when plugging the PM2800M Talkback Output into an RTS Program Input). Instead, an adaptor may be necessary.



CAUTION: Do not connect a standard XLR-3 cable between the PM2800M and the RTS (or similar) intercom system, except as noted above. It is essential that the proper isolation be provided, as explained below. Failure to observe this precaution may result in damage to the console and/or the intercom system. Besides, it simply won't work right.

To apply signal to the PM2800M COMM IN, signal can be derived from any RTS intercom XLR-3 line. Pin 2 of the RTS system carries the audio for channel 1, along with 32 volts DC to power the intercom units. Pin 3 carries the audio for channel 2 – it may or may not carry DC power. Pin 1 serves as the DC return and audio common. In order to block the DC from reaching the PM2800M, a 10 μ F, 50 volt capacitor should be installed in series with the line. As a precaution, this capacitor (or a second one) should be left in the line for channel 2. In fact, as a further precaution, capacitors should be installed on both primary and both secondary leads to the transformer, as this avoids a problem in the event that the adaptor is misconnected. The nominal line impedance for the RTS intercom system is 200 ohms, and nominal level is -7 dBu. In order to optimize the interface, RTS recommends the use of a 1:1 turns ratio (2k ohm) transformer. This transformer may have center taps which can be used to better match the -7 dBu RTS line level to the +4 dBu PM2800M line level. Of course, the PM2800M COMM IN is equipped with a gain control, so it not necessary to use a *center tapped* transformer. It is necessary to use a transformer for ground isolation. Be sure to use the isolation capacitors.

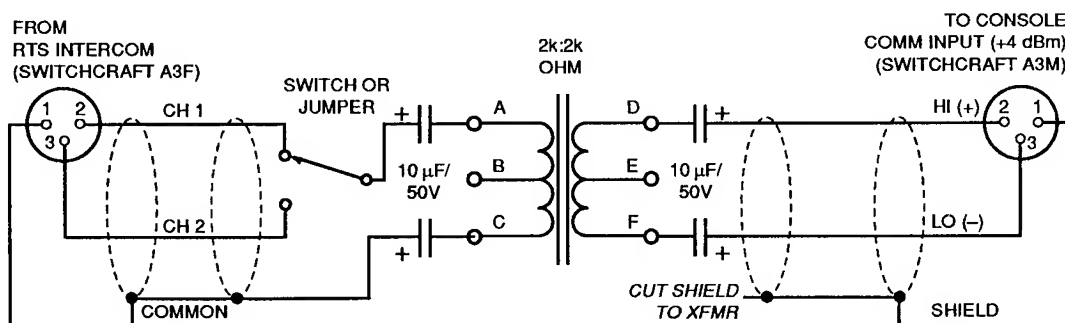


Figure 7-3. Interface of RTS Intercom to PM2800M “Comm In.”

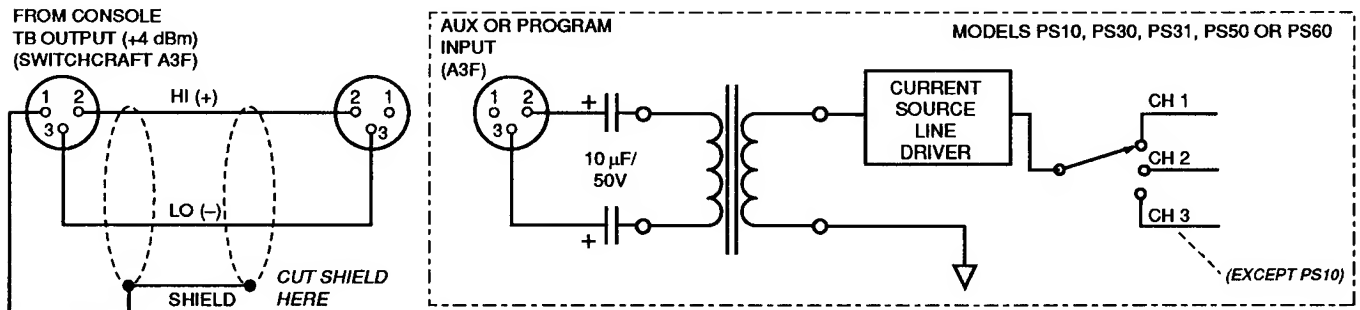
Assuming you do follow RTS' recommendation, we suggest using a mini-box to house the transformer and capacitor, equipping the box with clearly labeled XLR-3 connectors. If you use only one protection capacitor, then be sure you **DO NOT SWAP INPUT FOR OUTPUT**, as the capacitor protection will be lost, and the transformer will take the dc across its winding. Only one intercom channel can be fed to the console. The accompanying illustration depicts a jumper or switch to select the active channel, although if a "fixed" choice is made, then the appropriate conductor for channel 1 or 2 can be hard wired.

The PM2800M COMM IN Pad [83] should be set to "+4" position, and the LEVEL control [62] may be adjusted as required. You can obtain additional gain by wiring the transformer primary so that the audio from the intercom goes to the center tap (terminal B in Figure 7-3) instead of the high side of the winding (terminal A).

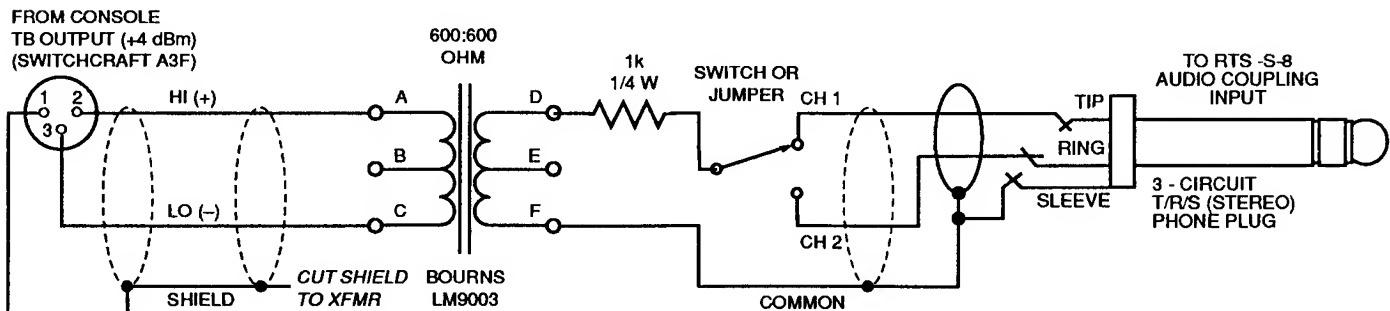
It is possible to communicate from the PM2800M to the intercom line. This is done in one of two ways: (1) via a special "Program" or "Aux In" connector on the

RTS Systems power supplies (on the Models PS10, PS30, PS31, PS50 or PS60), or (2) directly using an audio coupling input on the power supply (Model PS8 only). No blocking capacitors are necessary in this case because the coupling input is dc isolated within the power supply. However, RTS still recommends use of a 1:1 600 ohm transformer (again, the Bourns LM9003 or equivalent). In addition, RTS recommends placing a 1k ohm, 1/4 watt resistor in series with the circuit from transformer secondary to the intercom — both to drop the nominal +4 dBu console output level so it is closer to the -7 dBu nominal intercom line level and to lessen the loading of the intercom line.

RTS Systems power supply Models PS10, PS30, PS31, PS50, and PS60 have a balanced input connector (XLR 3-pin female) for the audio coupling input. The connector is wired as a pin 2/pin3 balanced input. The expected level is +4 dBu into 10 k ohm (a few older units may be 600 to 2000 ohms). A level adjustment and a channel assignment switch are located on the RTS Systems power supply. The interconnect is shown in Figure 7-4A. The smaller RTS power supply



A) For Larger PS10, PS30, PS31, PS50 or PS60 Power Supply



B) For Smaller, PS-8 Power Supply

Figure 7-4. Interface of PM2800M "TB Out" to RTS Intercom "Audio Coupling" Input.

(Model PS-8) is equipped with a 1/4" Tip/Ring/Sleeve phone jack for the audio coupling input. This jack is wired with the tip for channel 1, the ring for channel 2, and the sleeve for audio common. The interconnect is shown in Figure 7-4B. You have the choice of using a jumper, a switch, or wiring the connection to one or the other intercom audio channel. If the signal level is too "hot" for the intercom system, you can turn down the TB LEVEL in the console, or you can wire the secondary of the transformer so that signal is derived from the center tap (terminal "E" in Figure 7-4) rather than the high side of the transformer (terminal "D").

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7.3.2 Clear-Com Intercom Systems

Like RTS, Clear-Com also utilizes XLR -3 type connectors and a single shielded pair between their standard intercom stations so that ordinary microphone cables may be employed. Clear-Com differs from RTS in that standard Clear-Com equipment provides a single channel of communication on a standard XLR-3 microphone cable. Unlike RTS, standard Clear-Com equipment does not combine intercom audio with the DC operating voltage. Pin 2 of the XLR carries the 30V DC power — no audio. Like RTS, a Clear-Com intercom line cannot be plugged directly into the Yamaha console.



CAUTION: Do not connect a standard XLR-3 cable between the PM2800M and the Clear-Com (or similar) intercom system. It is essential that the proper isolation be provided. Failure to observe this precaution may result in damage to the console and/or the intercom system. Besides, it simply won't work right.

Interface Method N° 1

To connect the console directly to the Clear-Com intercom line, use the adapter circuit illustrated in Figure 7-5. The "Transmit Level" control is required to adjust the output level of the console to match the level of the Clear-Com line. The COMM LEVEL control on the console will adjust the level coming from the Clear-Com intercom line. With this method of interface, there is no "anti-sidetone" function. If the

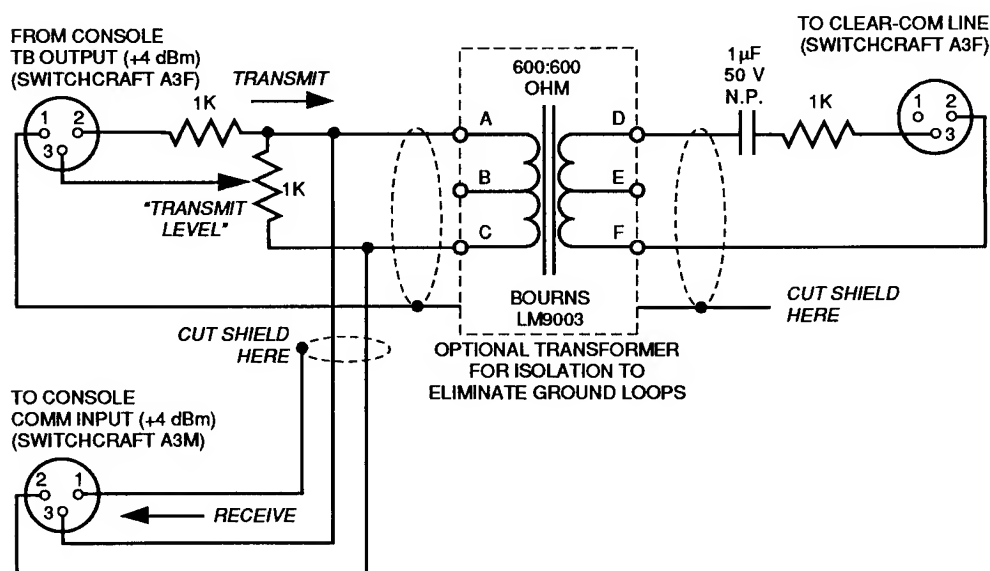


Figure 7-5. Adapter Cable for Direct Connection of a Clear-Com Intercom Line to the PM2800M "Comm In" and "TB Out".

operator is simultaneously listening and talking to the intercom line, his or her own voice will come back to his/her own ear at a very high level — possibly higher than the voices of others on the intercom line. This problem can be avoided by using an alternate interface method, N° 2.

Interface Method N° 2

To connect the console to the Clear-Com system through either a 4-wire to 2-wire interface (Clear-Com IF4-4 or AC-10K/H), or by using the headset jack on an intercom station, use the adapter circuit illustrated in Figure 7-6. The “Transmit Level” control is required to adjust the output level of the console to match the level of the Clear-Com line. The COMM LEVEL control on the console will adjust the level coming from the Clear-Com intercom line. If used with an intercom station, be sure to include the additional circuit shown within the dashed line; set the “Microphone On/Off” switch on the station to ON position, and adjust the station “Listen Volume” control to approximately the midpoint of its range. If used with an IF4-4 or AC-10K/H interface, the pad circuit (within the dashed line) should be deleted. This interface method provides the proper “anti-sidetone” function so the operator will hear the others on the line at a higher volume than his/her own voice.

Note: The preceeding information refers to standard Clear-Com single-channel intercom circuits. Clear-Com “TW” type circuits and beltpacks do combine one channel of audio with DC power. Contact the Clear-Com factory for special “TW” interfacing information.

For additional information, contact:

Clear-Com, Inc.
945 Camelia St.
Berkeley, CA
Phone (415) 527-6666
FAX (415) 527-6699

(Special thanks to Bob Tourkow of Clear-Com and to Stan Hubler of RTS Systems for technical assistance with this section of the manual.)

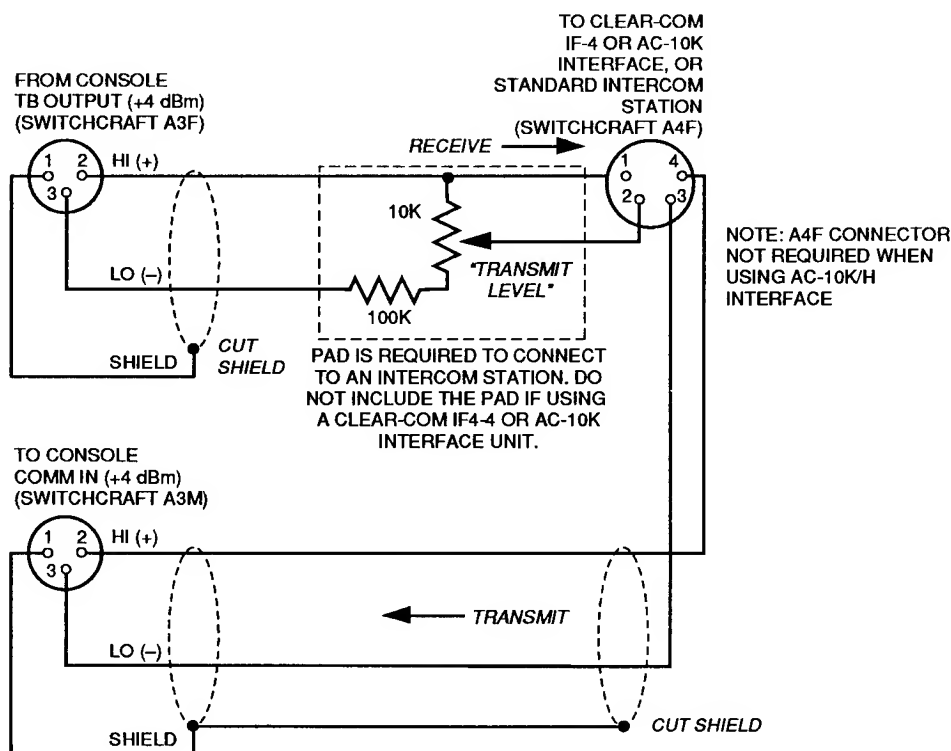


Figure 7-6. Adapter Cable for Connection of a Clear-Com Intercom Using IF4-4 or AC-10K/H Interface, or the Headset Connector of a Standard Intercom Station to the PM2800M “Comm In” and “TB Out”.

SECTION 8. APPLICATIONS

8.1 GENERAL

The PM2800M is designed primarily for stage monitor mixing in live sound reinforcement applications. Its exceptional flexibility, however, enables it to be used as an occasional general-purpose audio mixing console for other applications, including TV show and music video production, multi-media/A-V audio production, and general recording. We explain a few reasons why the PM2800M is useable in these applications below, but our primary focus is on stage monitor applications, for which the PM2800M is exceptionally well equipped. It is up to you, as the sound engineer or mixing console operator, to best utilize these capabilities in your specific application. This manual is by no means comprehensive, and we expect that many of you will devise unique means to connect and utilize the PM2800M. In fact, Yamaha encourages you to share your special applications with us so that we may, in turn, share the general concepts with other PM2800M users.

8.2 STAGE MONITORING

The PM2800M's electronically balanced inputs are of the highest quality, and input transformers can be installed internally where the extra isolation is required. Input channel sensitivity is now broadly adjustable from -80 dBu to -6 dBu by means of a 3-position attenuation Pad plus a Gain trim control with 34 dB range, so there's plenty of gain when it's needed, and noise is minimized when the extra gain is not needed. Four band sweep-type equalization, plus a sweep-frequency high pass filter, facilitate precise shaping of the sound.

Individual mix controls on each input module apply signal to any (or all) of the eight primary mix busses in whatever proportion the operator desires. There is a separate, directly-assignable stereo mix bus, which can be used as though it were two additional discrete mixes when stereo is not required. The primary (and stereo) mixes may be used to drive different banks of amplifiers and stage monitor speakers, each optimized for a particular performer (or performers). In addition, however, there are four aux sends per input, applying individually-adjustable signals to the four aux mixing busses. These four mixes may be used for echo/effects sends; if not all are needed for this purpose, the remaining busses may be used to create additional monitor mixes (for a total of up to 14 discrete output mixes).

The PM2800M goes a step beyond the traditional monitor mixing console in that it also includes a 10 x 4 Mix Matrix. The matrix may be used to remix any of the eight primary mixes and the stereo mix into four different mono (or two stereo) outputs. The matrix also may be used as a proportional distribution system, whereby a given mix is sent to more than one monitor amplifier/speaker system, with discrete control over the output. Thus, for example, should the lead vocalist and the bass player trade stage positions with each other, their monitor mixes need not be duplicated on two primary mix busses; each of their discrete mixes can be re-routed to the appropriate set of monitor speakers via the matrix – providing a continuous, uninterrupted flow of monitor sound. Thus, the true output capability of the console is far greater than the eight primary mixes, the stereo mix and the four aux mixes would suggest because the matrix avoids the need for “doubling up” of output mixes.

Corresponding to the four aux send busses, there are four auxiliary returns. Each is switchable to handle a mono or stereo signal. Moreover, the aux returns may be used as simple mono or stereo line inputs to the console. Consequently, input channels are not “used up” unnecessarily to handle echo/effects returns, tape recorded inputs, or remote signals fed from the house console.

The PM2800M is built to be both fast and precise. It is built for use by a person who has to make instant decisions that directly affect the performance of every person on stage, where an error can destroy the composure of the performers and the mood of an entire show. It is built to make it easier to avoid feedback, and, should feedback begin, to kill it before it destroys the show. The PM2800M is a toolbox of features and functions to keep the performers happy and the monitor mix engineer in firm control.

This is why the PM2800M is equipped with an extensive cue/solo system that has input priority logic – a system that enables the operator to monitor any combination of output mixes as a “normal” mode, yet instantly substitute any input in the headphone or local monitor speaker which the operator suspects of creating feedback or which might need EQ or level adjustments. (The console can even be linked to standard intercom systems so the operator doesn't have to wear two sets of headphones.)

A Signal present LED and two Clip LEDs per channel make it easier to prevent input channel levels from getting out of control, and extensive output bus metering make it easy determine whether a bus is too hot or perhaps has too little level for good signal-to-noise performance. Still, as useful as the metering is, it won't prevent feedback, so we have installed DIM switches that provide a fast, nearly foolproof way to kill incipient feedback by partially muting an output mix until the specific "problem" input to that mix can be located and corrected. Polarity reversal switches on each input provide the operator with an additional tool to fight feedback. What is more, the Insert In/Out jacks on each input and mix bus permit signal processors (noise gates, graphic equalizers, or compressor/limiters to name a few) to be used at the appropriate point, providing the needed control of the mix without over-processing signals that don't need it — and perserving optimum signal-to-noise performance.

The eight Mute Groups constitute a powerful tool to facilitate scene or set changes, and to optimize the stage for solos; by assigning an input to one or more mute groups, the operator can, with the touch of a finger on a Master Mute switch, turn on or off assigned groups of input channels. Overall control of the mix is excellent because there are linear faders on each input channel, as well as on the mix bus outputs. Faders provide immediate visual confirmation of level, and multiple faders can be adjusted simultaneously with a single hand (or forearm). Obviously, it is not practical to provide 10 or 14 faders per input (one for each channel's contribution to each mix bus), so Yamaha did the next best thing. We provided individual rotary controls for each of the 14 mixes (8 primary mix, 2 stereo mix, 4 aux send), plus a linear fader that can be used to control the overall contribution of the rotary controls to their respective busses. As shipped, the input modules are configured so that the aux and stereo mix controls derive signal pre-fader; however, internal jumpers permit them to be set post-fader. The eight primary mix controls are equipped with individual front-panel switches so each can be set pre or post fader, as the needs may dictate. This makes it easy, for instance, for the operator to vary the *actual level* of the signal from a particular mic to all mixes by moving a single fader — without affecting the relative *proportion* of that mic sent to the different mixes.

8.3 SOUND REINFORCEMENT HOUSE MIX

The eight primary mix busses can be used to sub-mix various vocal and/or instrumental sections, and these can be remixed to mono or stereo for the house feed by using the Mix Matrix. In this mode, the mix controls on each channel will be set to Post fader mode, and will be turned either all the way up ("assigned") or all the way down ("off"); the channel fader then adjusts the contribution to any assigned mix busses. Since there are another four auxiliary busses and the stereo bus, there is no shortage of effects sends or foldback (monitor) sends. The four auxiliary returns are each switchable to handle a mono or stereo signal, so the input channels are not "used up" unnecessarily.

The PM2800M has other useful features for general sound reinforcement, such as: numerous LEDs to display signal levels, an all aluminum shell that reduces weight substantially without sacrificing strength, a low profile that blocks fewer seats in the house while providing a good sight line to the stage, and an extensive input-priority cue system, plus a solo "in place" mode that mutes other channels for faster setup and faster troubleshooting during sound checks.

8.4 THEATRE

The PM2800M has features that enable it to do a creditable job for theatrical sound reinforcement. Its eight Master Mute groups, together with the eight Mute assign switches on each input module, enable all the sound sources for a given scene to be preset so they can be turned on or off at the press of a single switch. Since the console has up to 104 dB of gain (Channel In to Matrix Out), distant microphones and quiet speaking voices will cause no problems.

The console's Mix Matrix can be used as an assignable output mixer. Not unlike a lighting console in concept, the Mix Matrix permits up to 10 sources (the eight group busses and the stereo bus) to be remixed into four different output mixes. The matrix outputs can drive various primary speaker systems, effects speaker systems, as well as lobby, dressing room and other remote speakers. The inputs to the matrix can be mixed independently, as required, for each of the areas. If a simultaneous recording is needed, the matrix can be set to mix signals from ahead of the mix master faders, so the mix and stereo outputs can be used for independent multitrack and two-track tape recording mixes. A Communication input and Talkback output facilitate interface to intercom systems.

The console has a center master, so two operators can work conveniently to handle the show. Its low

profile means better sight lines from a high balcony. Its rugged construction means it can travel, reliably, along with the show.

8.5 PRODUCTION

Getting the basics of a soundtrack on tape while you're trying to mix sound for a live show can be a real challenge. The PM2800M simplifies the task by providing independent mix capability for the live sound requirements and the tape recording. You can create 18 different output mixes (eight groups, four aux mixes, a stereo L & R mix, and four matrix mixes). With four aux returns, each of which accepts mono or stereo sources, the input channels are not "used up" just to handle extra effects returns or pre-recorded cues. All channel inputs, aux returns, and bus outputs are balanced, low impedance circuits, enabling long cables to be used without excess noise; optional transformers are available where the extra requirements for grounding isolation or high common mode voltage are critical.

Eight mix master faders and eight Master Mute groups together enable the console operator to more easily "keep track" of the many inputs, switching them on or off, and adjusting their levels at the touch of a finger... precisely on cue. Speaking of which, an extensive cue system, with input priority, enables any output or input to be scrutinized "in place" without affecting the output signals. A solo mode, which mutes all but the selected input, speeds pre-production setup and troubleshooting.

An important feature of the PM2800M for a production environment is the 10 x 4 mix matrix, a built-in "mixer within a console." In video work, for example, discrete output mixes can be fed to the 8-track tape machine from the mix outputs at a suitable level to maintain an ideal S/N ratio while avoiding tape saturation. At the same time, the mix matrix can create working mixes of those groups, with levels adjusted for more "listenable" reference monitoring or foldback. Alternately, some of the aux mix busses can be used for performer cue mixes or foldback, while others can be used for effects sends or to supplement the primary mixes whenever more tracks must be recorded. The eight mix outs plus four aux outs and two stereo outs fill up 14 tracks, and that leaves one spare plus a time code track on the 16-track recorder.

Built-in talkback and communication (intercom link) capability make it easy for the production personnel to coordinate efforts, and the console operator doesn't have to wear two sets of headphones. In fact, the cue output jacks make it possible to monitor the console outputs without any headphones.

8.6 POST PRODUCTION

Once a show has been photographed on video, film or multi-image media, it's time for the crucial post production job of mixing sound effects, music, and/or dialog. Sometimes there is no "original" production soundtrack, and all recording is done in the post production phase, while other times the post production task is primarily one of enhancement. In any case, the PM2800M can do the job. Its many inputs can be switched to handle virtually any input level, from the lowest level mics to very "hot" electric guitars, electric keyboards, and virtually any tape recorder or film chain. Cue switches on just about every input and bus make it possible to check signals "in place" without disrupting the output mixes. Sounds can be precisely tailored, and defects "surgically removed" using the four-band sweep-type equalizers on each input channel, as well as the sweep frequency high pass filters that go as high as 400 Hz. Insert in/out jacks on every input channel and the aux busses make it possible to patch in whatever signal processing is desired. Added convenience is provided by the eight Master Mute groups, which switch assigned input channels on and off instantly.

A mix matrix permits the eight groups and the stereo bus to be mixed into four different outputs. This 10 x 4 matrix, a "mixer within a console," makes it possible to control groups of similar instruments (or vocals) with the mix fader, and to then remix those groups. In film work, for example, the mixes might be: left, center, right, surround. If the matrix is fed pre-mix fader (internal jumpers permit this), then mixes 1 through 8 can be used to create up to four different stereo mixes (odd left/even right, using the individual mix assignment controls on each channel in lieu of pan pots); the stereo mix itself can serve as a fifth stereo mix. Within the matrix, three of these independent stereo mixes can be combined into a reference stereo mix. This could be useful, say, for stereo music, stereo dialogue and stereo effects, plus an overall monitor mix. Cue output jacks, in addition to the headphone output, make monitoring more convenient. There is also a communication input and talkback output that facilitate interface to intercom systems.

8.7 VIDEO

In today's highly active video production scene, more live music, more pre-recorded sources, and more special effects are being applied to create soundtracks to which people are paying more attention than ever. Stereo VCRs, stereo TV broadcast, and the tremendous success of the "music video" have placed very strong emphasis on video sound quality. With its high quality sound and powerful capabilities, the PM2800M can be used to satisfy many video sound production requirements. The console's 32 or 40 input channels can handle the numerous mics, instruments and pre-recorded sources for almost any production... and sub inputs allow two consoles to be linked together for that once in a great while when even more inputs are needed. (In fact, we foresee one application for the PM2800M will be to add extra inputs to a system in which a PM1800 or PM3000 is the primary mixer, but more channels and submixes are needed for a particularly complex production.)

The PM2800M has eight primary mix busses, so different groups of instruments or mics can be assigned to their own group and controlled with a single fader. All eight of these can be used for submixes, and reassigned in the matrix to create a master stereo mix. The PM2800M also has four auxiliary mixing busses that can be used for effects sends, for headphone cue mixes, or for additional subgroups. The mix matrix can be used to create monitor mixes of the various groups, while other console outputs simultaneously provide different mixes for recording. Cue output jacks can be used to feed local monitor speakers, and an input priority cue system lets the operator instantly check any input channel or auxiliary return at the touch of a single switch.

With four auxiliary sends, and four aux returns, it's easy to utilize the most sophisticated effects. The aux returns can each be used for a mono or stereo source. If even more returns are needed, input channels may be used (they have the added advantage of four-band sweep-type equalization). Built-in talkback and communication (intercom link) capability make it easy for the producer, director and crew to coordinate efforts, and the console operator doesn't have to wear two sets of headphones.

8.8 DETAILED STAGE MONITOR SETUPS, HINTS & TIPS

Several of the following subsections include large "wiring diagrams" which depict the PM2800M-32 rear panel and cables which are routed to and from various signal sources, signal processors, and so forth. Most of the diagrams utilize the identical list of sources (mics and instruments), typical of a pop music band. This is not to suggest that only one type of source is appropriate, but rather to keep one portion of the diagram "stable" so that changes in the signal processing and output routing will be more obvious. If you have other sources, or a 4-input console, the setups will vary, but the principles of operation will be the same. We have also labeled certain outputs as "Outputs to Amps & Speakers." In fact, compressor/limiters, and electronic crossovers for biamplication, may also be part of the equipment which follows these console outputs.

Note: We have used a oval-shaped dashed line to indicate a group of related cables in these diagrams. The dashed lines DO NOT indicate a shield around the cable or a snake. In fact, where we show input and output cables grouped together, this grouping is provided only for convenience of labeling; normally the cables for input to and output from a device should be physically separated to avoid crosstalk and the potential for high-frequency oscillation or electrically-coupled feedback.

8.8.1 Basic Stage Monitor System

In this setup, we are using the 8 primary Mix busses to drive various sets of power amplifiers and stage monitor speakers. Mix busses 7 & 8 are set up to provide a two-channel mix for the sidefill monitors (the pair of opposing speakers on either side of the stage aimed at the lead vocalist). This may be a stereo mix or a dual-mono mix. The stereo bus provides a two-channel stereo mix for keyboard player. Another stereo output is provided from the Cue system, although this may more often be a dual-mono signal depending on how the cue system is operated.

The four Aux Out connectors are each routed to a separate, single-channel signal effects device (perhaps a reverb, a delay line, a phaser, or a multi-effects unit like the Yamaha SPX-90). The mono outputs of these four devices are brought back into the four Aux Return inputs. As you know, each Aux Return input consists of two connectors, allowing for stereo sources, but since each of these is a mono return, only the L/Mono input connectors are used here. (The front-panel MONO switch must be engaged for the aux returns, too.)

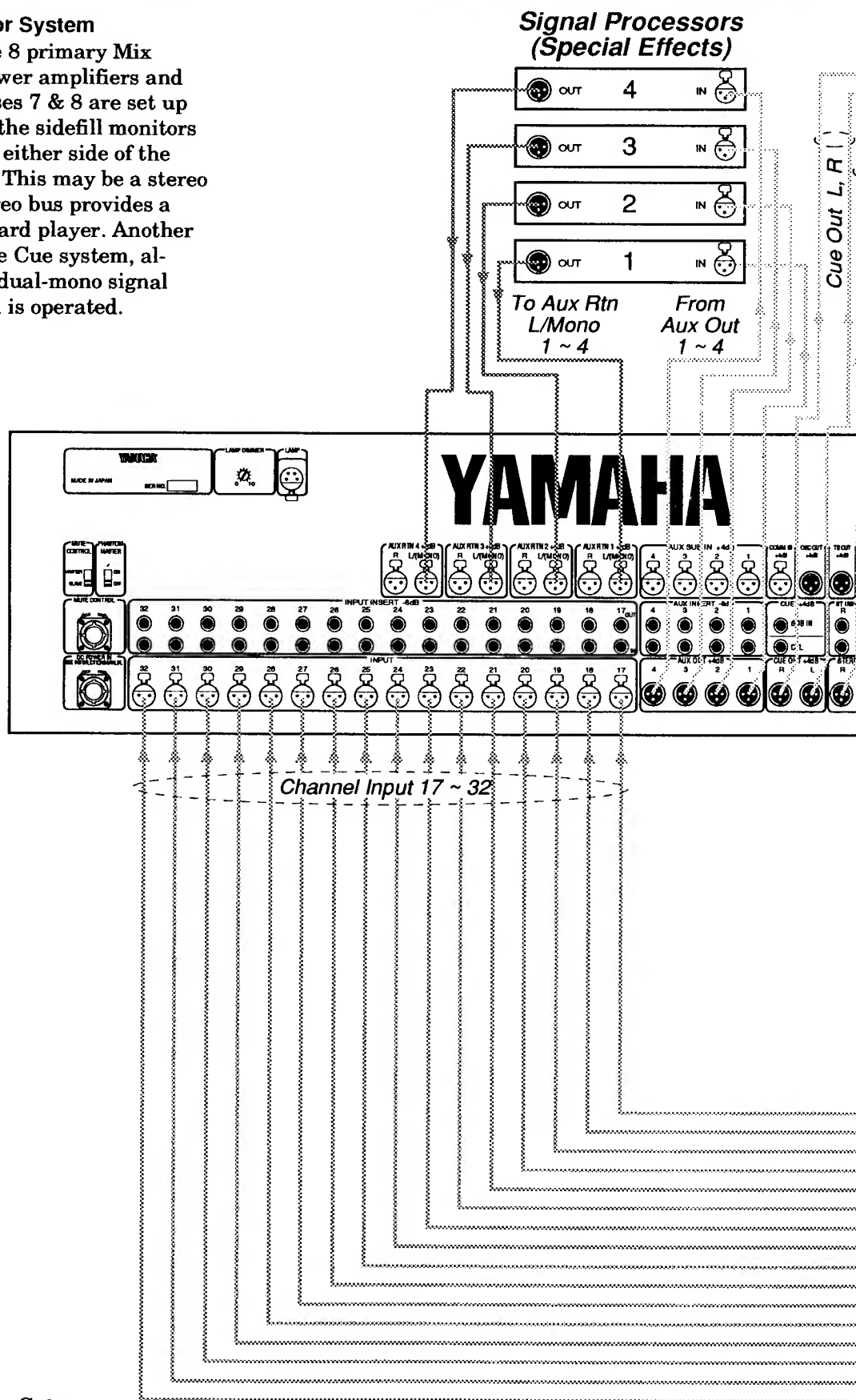
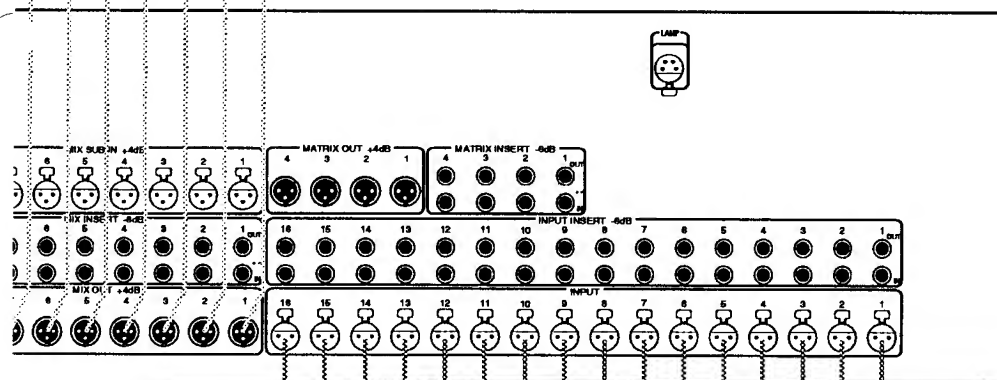
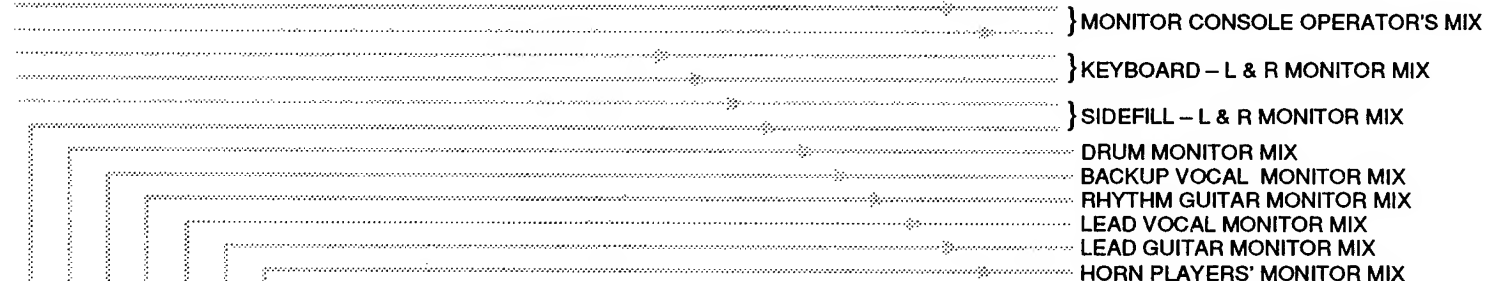
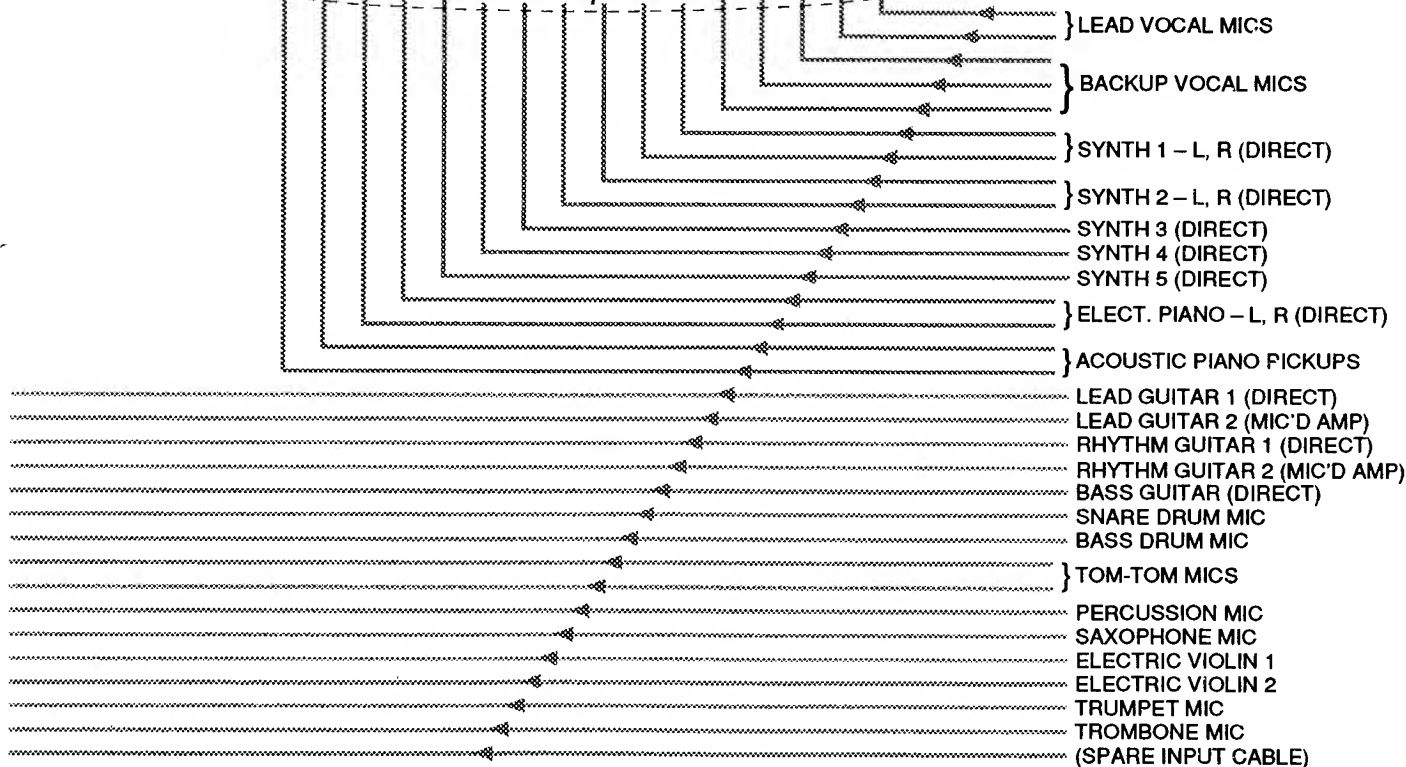


Figure 8-1. Basic Stage Monitor Setup

Outputs to Compressors, Amps & Speakers



Input Sources



8.8.2 Mono Input/Stereo Output Signal Processors

This setup is nearly identical to the prior setup, with the exception of the signal processors. In this case, each processor has a mono input, but a stereo output. This is typical of a number of reverb units, where one signal excites the reverb, and two outputs are extracted. Effects units such as the Yamaha SPX-90 also have this capability. In this case, only two of the four Aux Send outputs (3 & 4) are required, and two of the Aux Return inputs are used (1 & 2) – but in Stereo mode. Therefore, the front-panel MONO switches are disengaged on the two Aux Return channels being used (1 & 2). Aux Send outputs 1 & 2 are not needed to drive the effects units, these aux mixes are instead used to create additional program mixes, in this case to feed a stereo (or two mono) remote locations such as a broadcast van, the house console, etc.

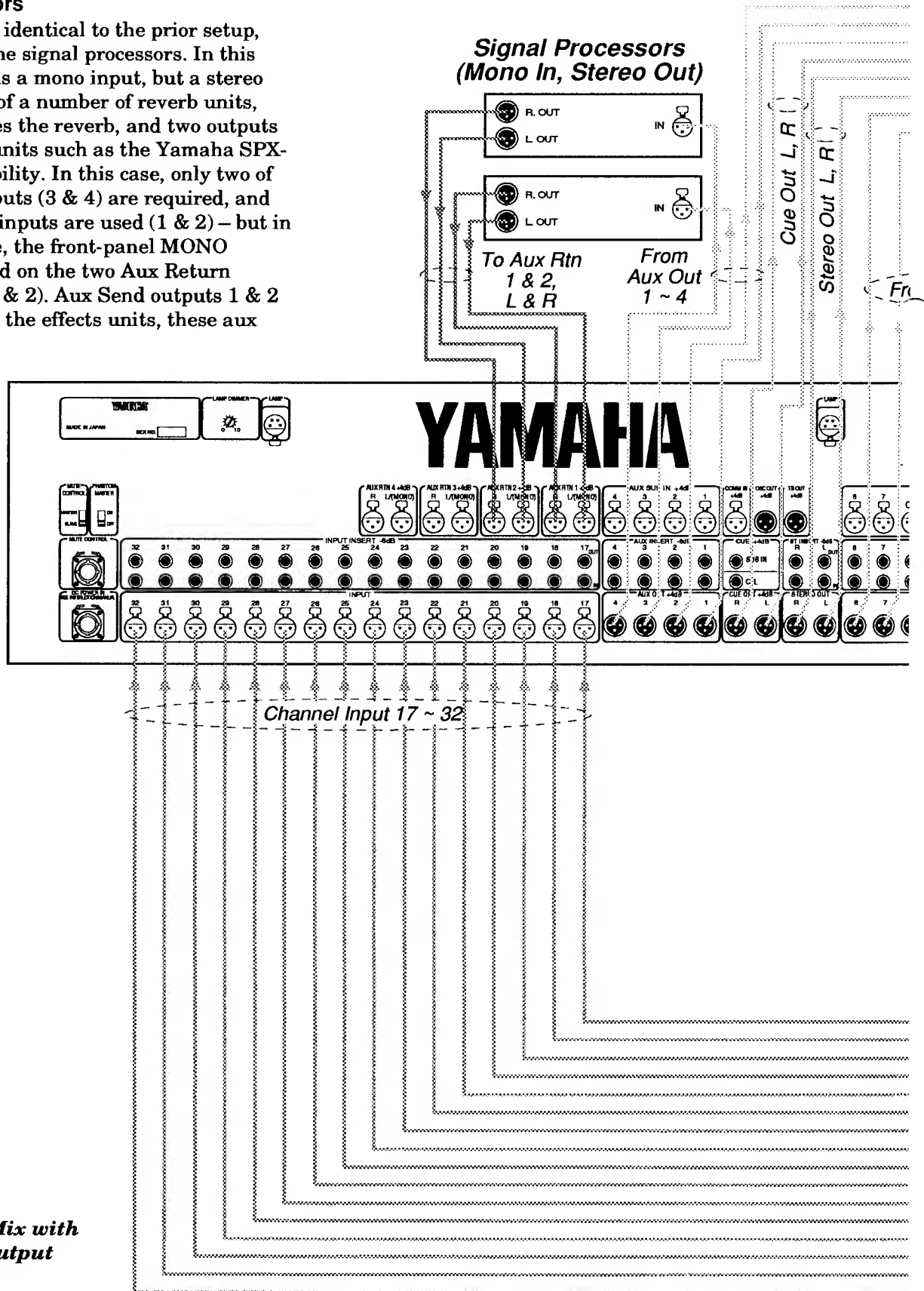


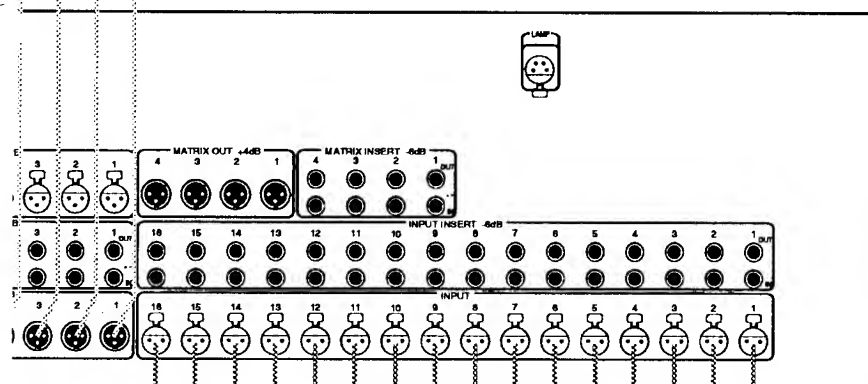
Figure 8-2. Monitor Mix with Mono Input/Stereo Output Signal Processors

Outputs

- } STEREO OR DUAL MONO MIXES FOR REMOTE LOCATION
- } MONITOR CONSOLE OPERATOR'S MIX
- } KEYBOARD – L & R MONITOR MIX
- } SIDEFILL – L & R MONITOR MIX
- DRUM MONITOR MIX
- BACKUP VOCAL MONITOR MIX
- RHYTHM GUITAR MONITOR MIX
- LEAD VOCAL MONITOR MIX
- LEAD GUITAR MONITOR MIX
- HORN PLAYERS' MONITOR MIX

to
Compressors,
Amps &
Speakers

Out 1 ~ 8



Channel Input 1 ~ 16

Input Sources

- } LEAD VOCAL MICS
- } BACKUP VOCAL MICS
- } SYNTH 1 – L, R (DIRECT)
- } SYNTH 2 – L, R (DIRECT)
- SYNTH 3 (DIRECT)
- SYNTH 4 (DIRECT)
- SYNTH 5 (DIRECT)
- } ELECT. PIANO – L, R (DIRECT)
- } ACOUSTIC PIANO PICKUPS
- LEAD GUITAR 1 (DIRECT)
- LEAD GUITAR 2 (MIC'D AMP)
- RHYTHM GUITAR 1 (DIRECT)
- RHYTHM GUITAR 2 (MIC'D AMP)
- BASS GUITAR (DIRECT)
- SNARE DRUM MIC
- BASS DRUM MIC
- } TOM-TOM MICS
- PERCUSSION MIC
- SAXOPHONE MIC
- ELECTRIC VIOLIN 1
- ELECTRIC VIOLIN 2
- TRUMPET MIC
- TROMBONE MIC
- (SPARE INPUT CABLE)

8.8.3 Graphic Equalizers on Primary Mix Busses, Tape Recorded Effects

Following a logical progression, we have taken the previous setup and further enhanced it. You therefore see the same pair of mono-input/stereo-output effects devices. In this instance, we noticed that two of the four Aux Return channels were not being used at all, so we put one of them to a different use. We patched the output of a stereo tape machine into the Aux 3 (L & R) inputs; this may be a cassette player or an open reel deck which contains pre-recorded effects or musical material. The Aux Inputs, after all, are standard line inputs, and this use enables the tape machine input to be accommodated without sacrificing valuable channel inputs.

In addition, we have connected four dual-channel graphic equalizers to the console. These are connected via the Mix Insert patch points (the Insert Out & Insert In jacks) so there is discrete control of the spectral balance for each mix. That is, if the localized response peaks in the lead vocalist's monitor must be reduced to avoid feedback, these spectral changes do not affect the sound being heard by the rhythm guitar player.

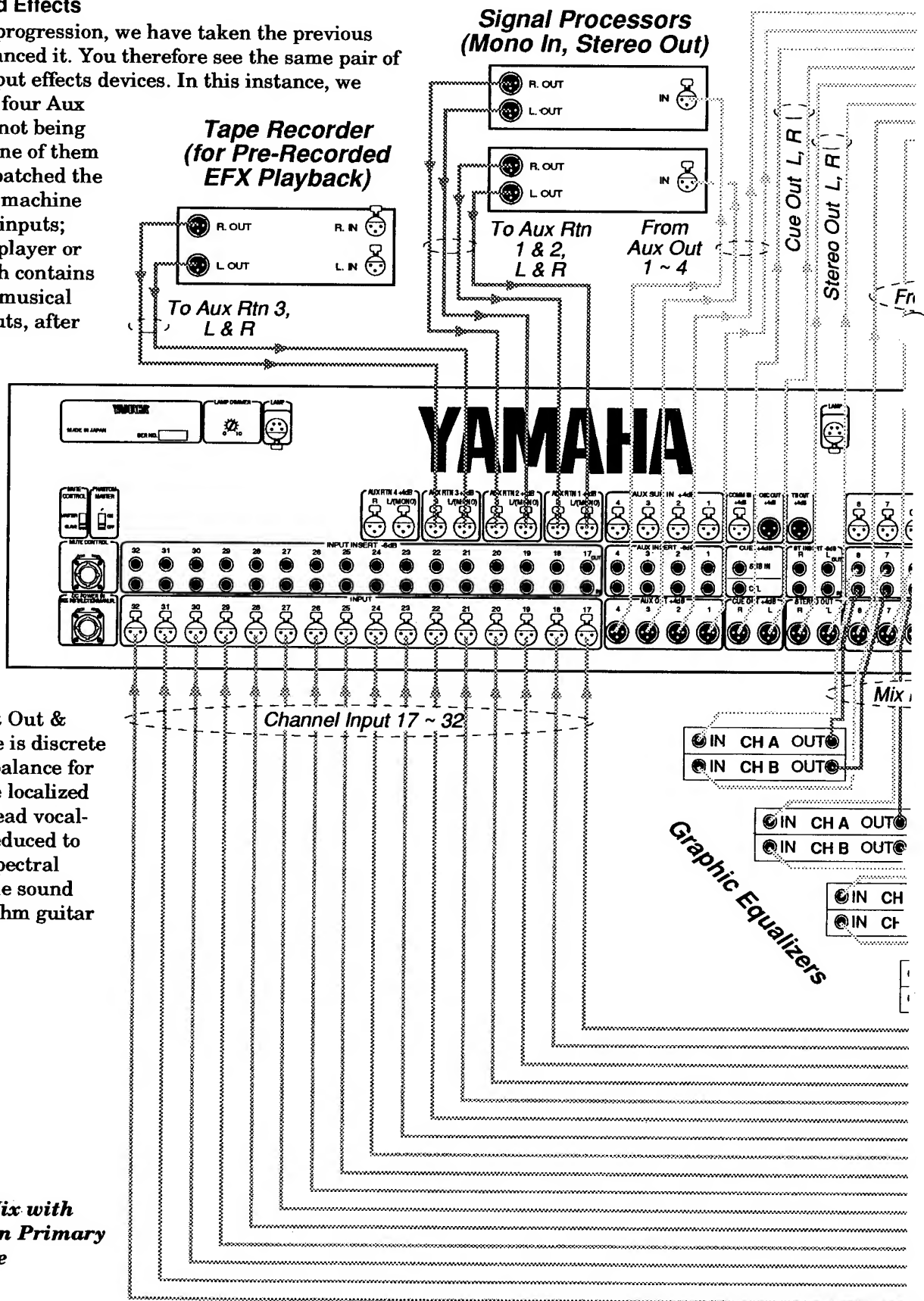


Figure 8-3. Monitor Mix with Graphic Equalizers on Primary Mix Busses; also, Tape Recorded Effects

8.8.3 Graphic Equalizers on Primary Mix Busses, Tape Recorded Effects

Following a logical progression, we have taken the previous setup and further enhanced it. You therefore see the same pair of mono-input/stereo-output effects devices. In this instance, we noticed that two of the four Aux Return channels were not being used at all, so we put one of them to a different use. We patched the output of a stereo tape machine into the Aux 3 (L & R) inputs; this may be a cassette player or an open reel deck which contains pre-recorded effects or musical material. The Aux Inputs, after all, are standard line inputs, and this use enables the tape machine input to be accommodated without sacrificing valuable channel inputs.

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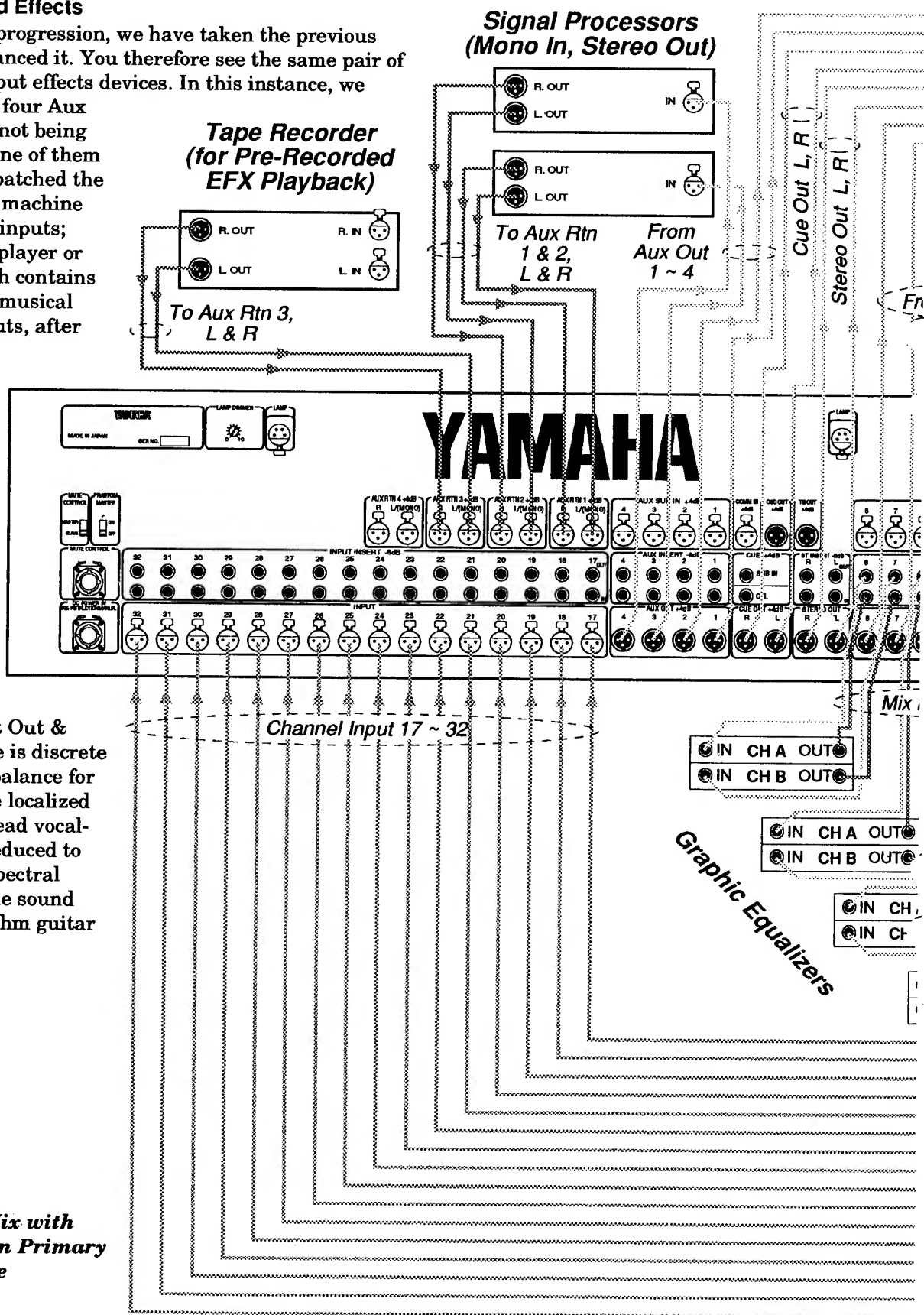


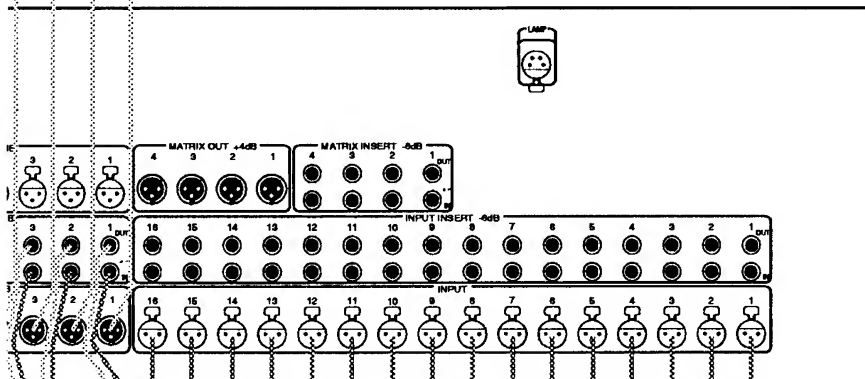
Figure 8-3. Monitor Mix with Graphic Equalizers on Primary Mix Busses; also, Tape Recorded Effects

Outputs

- } STEREO OR DUAL MONO MIXES FOR REMOTE LOCATION
- } MONITOR CONSOLE OPERATOR'S MIX
- } KEYBOARD – L & R MONITOR MIX
- } SIDEFILL – L & R MONITOR MIX
- DRUM MONITOR MIX
- BACKUP VOCAL MONITOR MIX
- RHYTHM GUITAR MONITOR MIX
- LEAD VOCAL MONITOR MIX
- LEAD GUITAR MONITOR MIX
- HORN PLAYERS' MONITOR MIX

to
Compressors,
Amps &
Speakers

Out 1 ~ 8



It/In

Channel Input 1 ~ 16

Input Sources

- } LEAD VOCAL MICS
- } BACKUP VOCAL MICS
- } SYNTH 1 – L, R (DIRECT)
- } SYNTH 2 – L, R (DIRECT)
- SYNTH 3 (DIRECT)
- SYNTH 4 (DIRECT)
- SYNTH 5 (DIRECT)
- } ELECT. PIANO – L, R (DIRECT)
- } ACOUSTIC PIANO PICKUPS
- LEAD GUITAR 1 (DIRECT)
- LEAD GUITAR 2 (MIC'D AMP)
- RHYTHM GUITAR 1 (DIRECT)
- RHYTHM GUITAR 2 (MIC'D AMP)
- BASS GUITAR (DIRECT)
- SNARE DRUM MIC
- BASS DRUM MIC
- } TOM-TOM MICS
- PERCUSSION MIC
- SAXOPHONE MIC
- ELECTRIC VIOLIN 1
- ELECTRIC VIOLIN 2
- TRUMPET MIC
- TROMBONE MIC
- (SPARE INPUT CABLE)

A OUT
B OUT

8.8.4 Compressor/Limiters on Individual Channels, Combination of Signal Processors

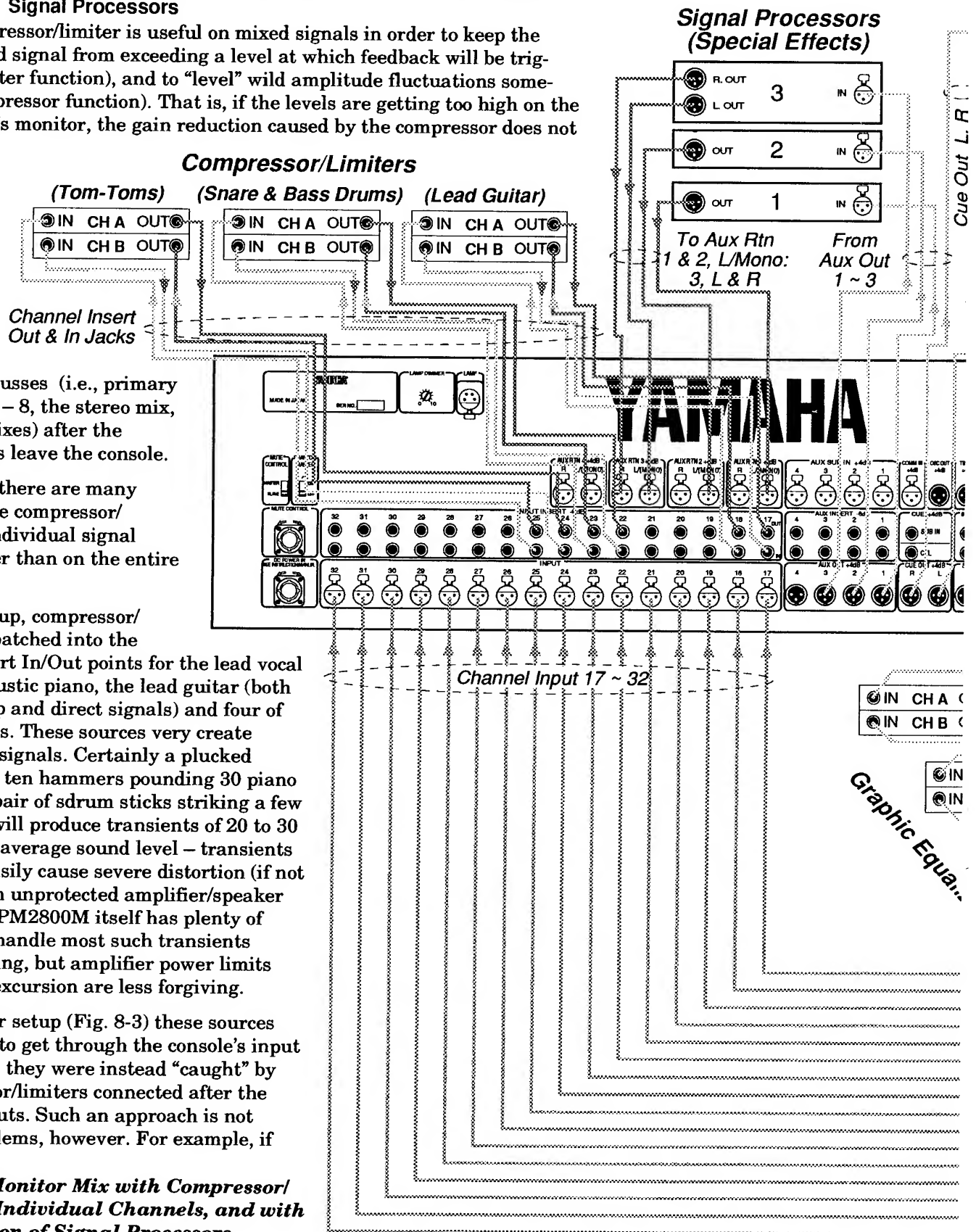
The compressor/limiter is useful on mixed signals in order to keep the overall mixed signal from exceeding a level at which feedback will be triggered (a limiter function), and to “level” wild amplitude fluctuations somewhat (a compressor function). That is, if the levels are getting too high on the lead vocalist's monitor, the gain reduction caused by the compressor does not affect the levels being heard by the rhythm guitar player. Typically, such compression is applied to the various busses (i.e., primary mix busses 1 – 8, the stereo mix, or the aux mixes) after the mixed signals leave the console.

However, there are many reasons to use compressor/limiters on individual signal sources rather than on the entire mix.

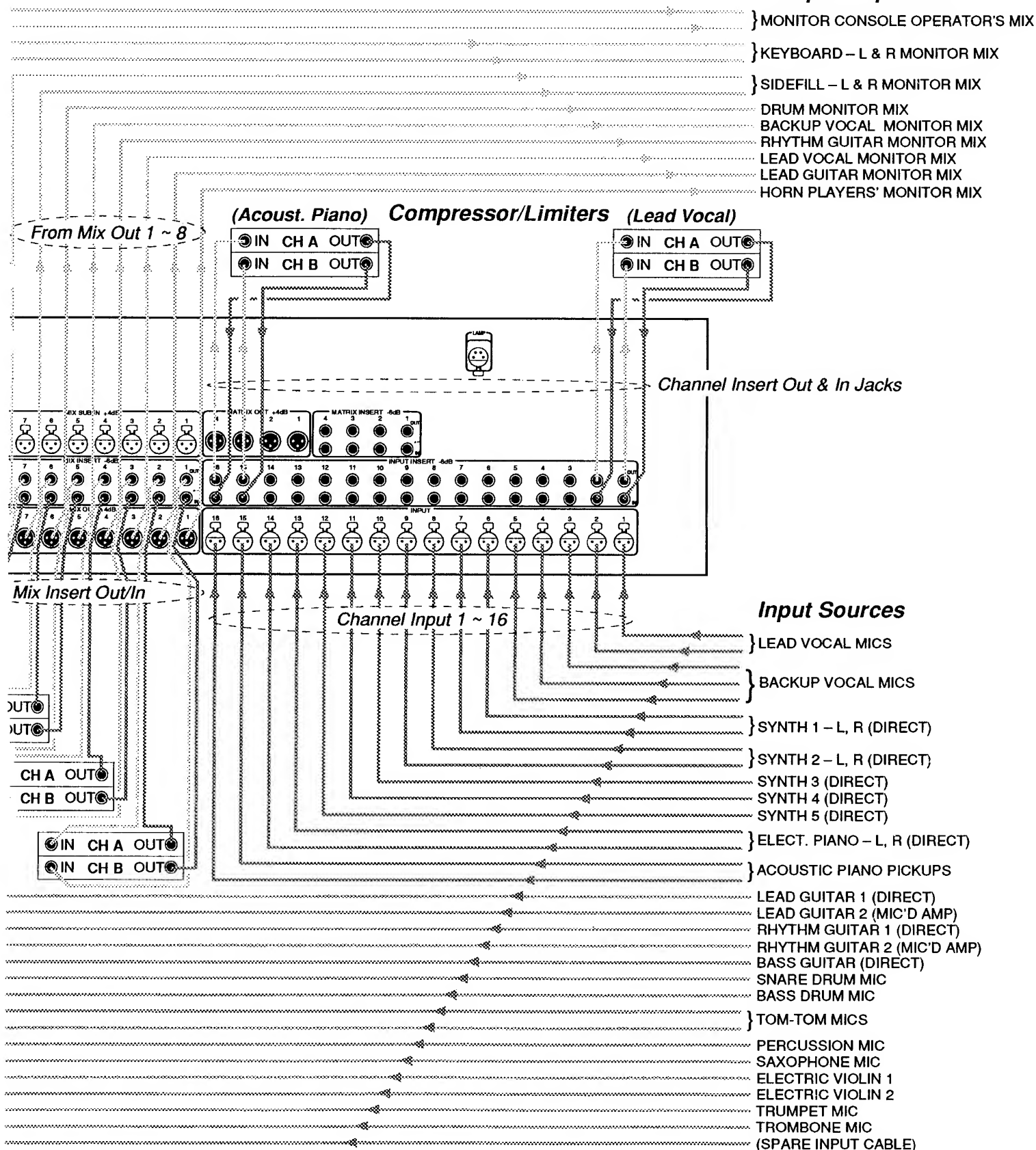
In this setup, compressor/limiters are patched into the Channel Insert In/Out points for the lead vocal mics, the acoustic piano, the lead guitar (both the mic'd amp and direct signals) and four of the drum mics. These sources very create sudden, loud signals. Certainly a plucked guitar string, ten hammers pounding 30 piano strings, or a pair of sdrum sticks striking a few drum heads will produce transients of 20 to 30 dB above the average sound level – transients that would easily cause severe distortion (if not damage) to an unprotected amplifier/speaker system. The PM2800M itself has plenty of headroom to handle most such transients without clipping, but amplifier power limits and speaker excursion are less forgiving.

In the prior setup (Fig. 8-3) these sources were allowed to get through the console's input channels, and they were instead “caught” by the compressor/limiters connected after the Mix Bus outputs. Such an approach is not without problems, however. For example, if

Figure 8-4. Monitor Mix with Compressor/Limiters on Individual Channels, and with a Combination of Signal Processors



Outputs to Compressors, Amps & Speakers



the keyboard player has some lead vocal and some drum sound in his monitor mix, and there are not compressors in the input channels, then the keyboard monitor mix's compressor/limiter will "duck" the overall mix ("duck" in this context means "force the level lower") every time there is a strong drum hit or a particularly loud shout from the vocalist. Such a setup would cause the keyboard player's monitor levels to move up and down unnecessarily, and inappropriately, because too much gain control is being applied to the *mixed* signal. Instead, by using the setup shown here, the drums are handled by four channels of compressor/limiter so that a loud tom-tom hit does not "duck" the bass drum (or vice-versa), and none of these will "duck" the keyboard player's monitor levels. Similarly, loud shouts won't affect the drummer's or keyboard player's monitors, nor will forte piano crashes force down the level of other performers' sounds in the various mixes.

In general, the quality of the sound will be better if compression (or limiting) is applied to individual input sources rather than mixed (bus) signals. Still, compression on the busses is helpful to catch problems that slip through from "unprotected" (not compressed) input channels, and to prevent cumulative levels from occasionally adding up "just the wrong way" to produce a too-high bus level even though the average and peak levels from individual sources are acceptable.

Another variation in this diagram illustrates the use of three signal processors fed by three of the Aux Out jacks: one is a mono in/stereo out processor, and two are mono in/mono out processors. As a result, three of the four Aux Return channels are used for signal processor returns, and the fourth is not used (it could be used as a mono or stereo line input for a tape machine, a line-level keyboard output, etc.)

8.8.5 Utilizing the Mix Matrix to Reroute Signals on Stage

In this setup, we have brought the PM2800M Mix Matrix into use. In this particular band, the lead vocalist, lead guitar player, rhythm guitar player, and sax player occasionally trade positions (perhaps trade instruments, perhaps not) on stage. With a conventional monitor mixing console, separate mixes would have to be created for each performer at each position – with the added complication of assignment of the mixes to the available amps and speakers. The PM2800M Mix Matrix makes the job much easier.

Four mixes are created in the conventional manner, using a different primary mix bus for each of these performers. However, instead of feeding the Mix Outs directly to the monitor amp/speaker systems, these four mixes are fed to the matrix, and the Matrix

Outs instead feed the amps and speakers for each of the four on-stage positions. Then, if a performer moves to another position, his mix can be turned down in the matrix strip where he had been standing and turned up in the new position. Similarly, the other performers's mixes can be reassigned in the matrix to suit their position on the stage. New mixes are not required, and mix knobs on the input channels are not “used up” unnecessarily.

This same approach to reassignment could also be used for a keyboard player who has two different keyboard setups and moves back and fourth between them. We do not want to have the “unoccupied” keyboard position's monitors turned on when the keyboard player is standing at the other position. In a conventional system, it would be necessary to devote two sets of stereo mix busses to the keyboards (with

See Figure 8-5 on next page.

duplicate mixes) so that one stereo output could be turned on and the other off, thus "using up" four mix knobs on each of the input channels which contribute to the keyboard mix. The Mix Matrix makes that approach obsolete.

Instead, the keyboards are mixed on one stereo bus; the output of this bus (L & R busses) is not fed to amps and speakers, but instead through the Mix Matrix (the ST to MATRIX PUSHBUTTON [53] must be engaged if the Stereo Bus is used for the keyboard mix). Two of the matrix strips are used for L & R outputs at one keyboard location, and the remaining two matrix strips for the stereo mix at the other keyboard location. Then the appropriate matrix outputs can be switched on and off [41], or (for smoother transitions) the Matrix Masters [39] can be cross-faded.

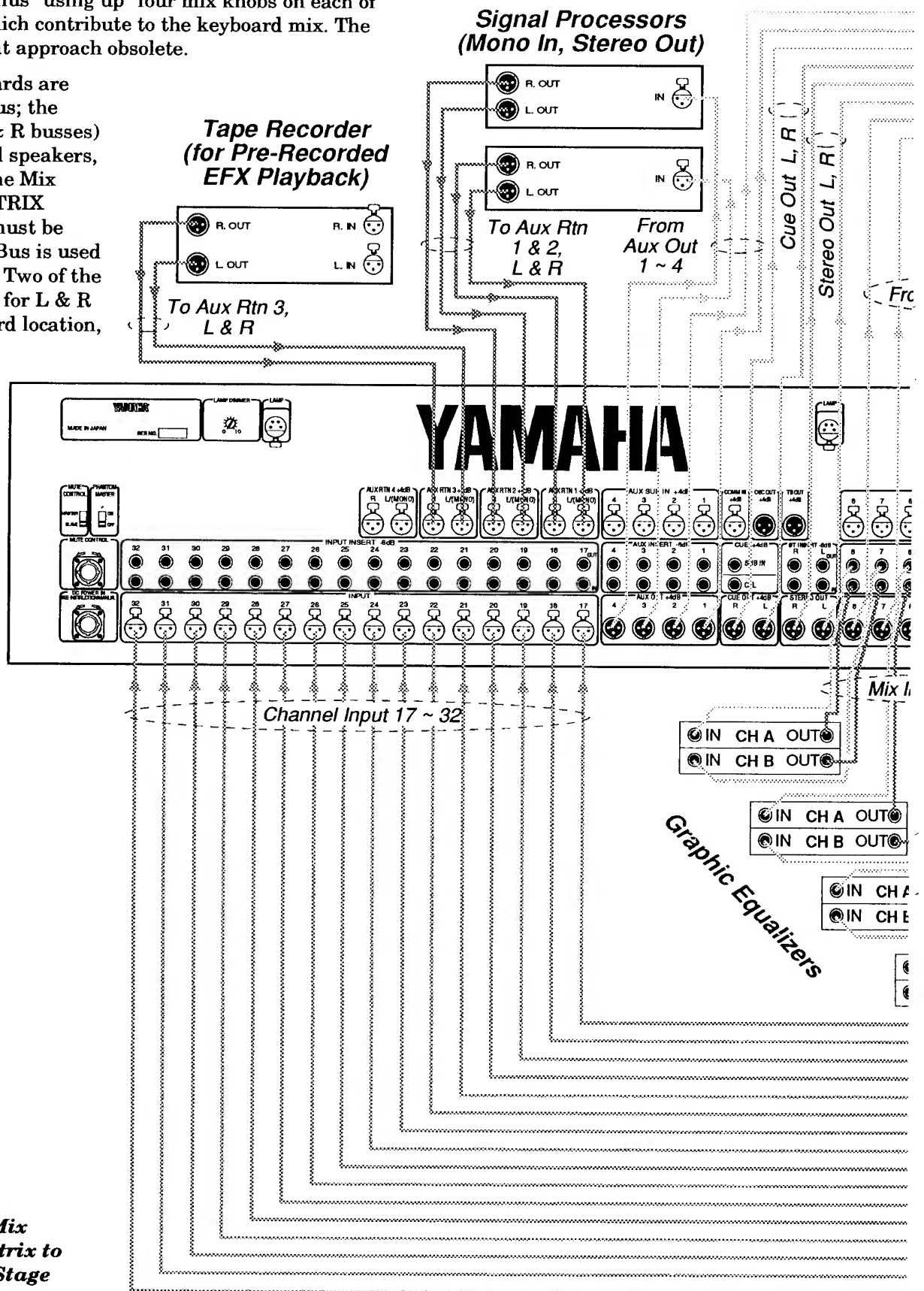
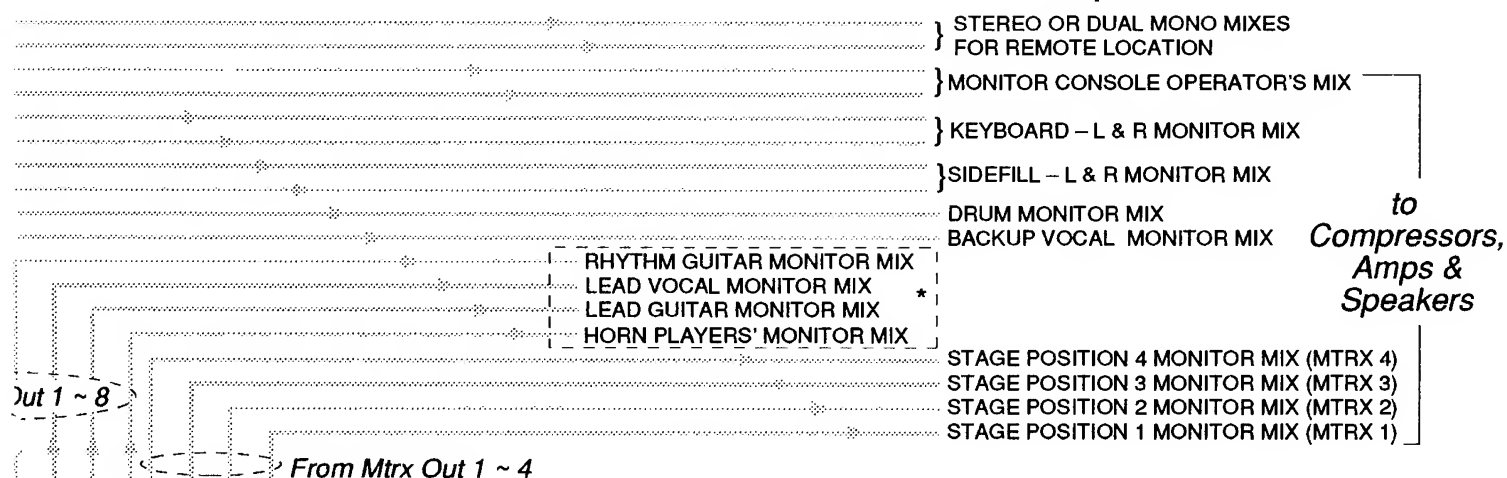
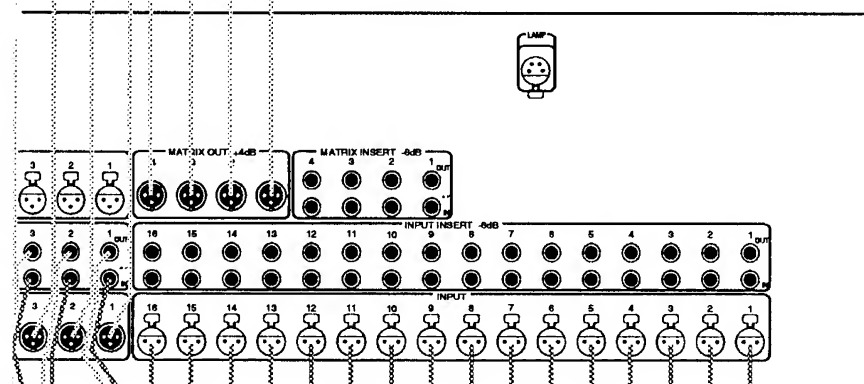


Figure 8-5. Monitor Mix Utilizing the Mix Matrix to Reroute Signals On Stage

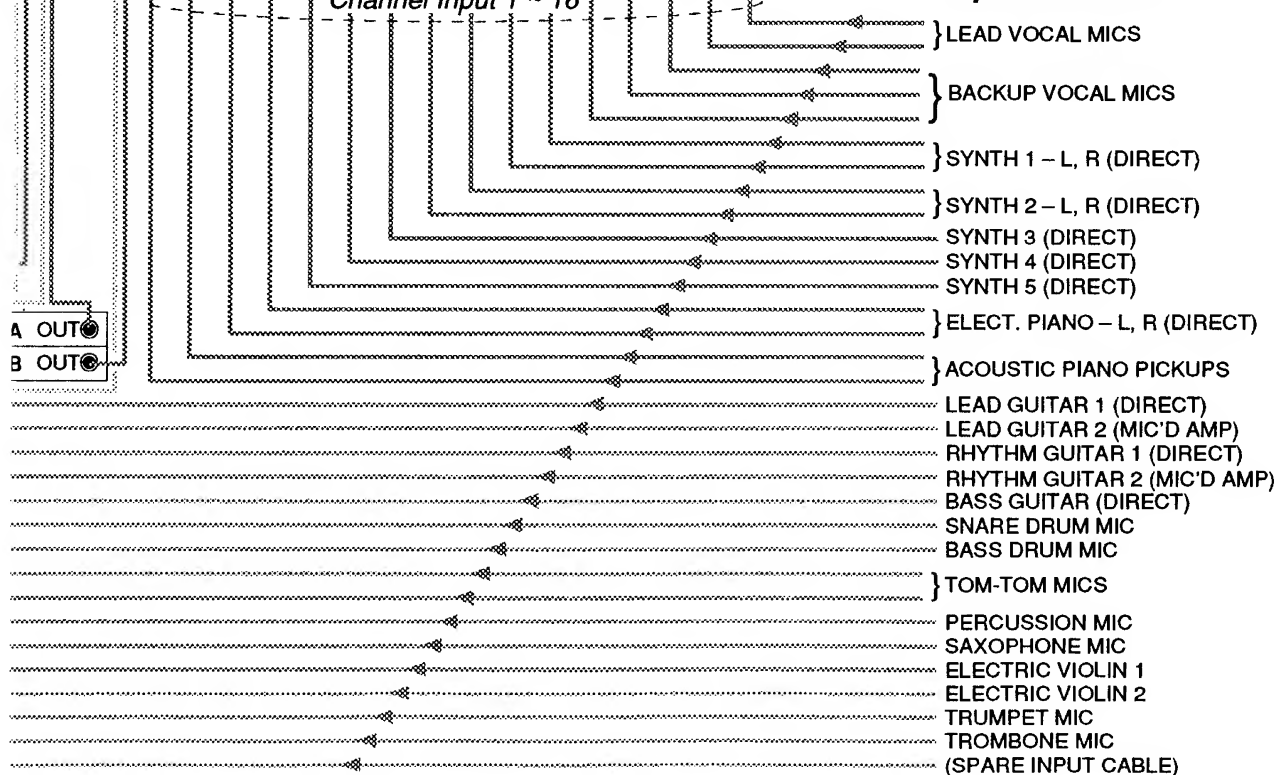
Outputs



* These mixes are not fed directly to amps & speakers, but instead are remixed in the matrix so they can follow performers as they move around the stage and/or trade positions



Input Sources



8.8.6 Additional Mix Matrix Applications

In live sound reinforcement, instead of feeding the house sound system directly from the Mix outputs [105], or the Stereo output [104], the sound system can be fed from the Matrix outputs [82]. In this case, the Mix busses are used for mixing sub-groups of different sources; i.e., brass, drum/percussion, lead vocals, backup vocals, rhythm guitars & bass, lead guitar, keyboards (in stereo), and so forth. The Mix Faders [35] then control the overall level of each sub-mix of input channels. The matrix channels can be used to create two stereo or four mono mixes from those mixes. The mix matrix outputs then feed the power amps and speakers for various zones in the main house, as well as other areas (dressing rooms, lobby, remote feeds, etc.)

The advantages to this approach are numerous. For example, if the brass level is too high in all outputs, only one Mix Fader [35] need be adjusted (for the brass subgroup). On the other hand, if there is too much vocal near the front of the audience (due to spill from the vocal stage monitors), you can adjust the one matrix mix level control [37], corresponding to the vocal Mix, in the matrix channel that feeds the near stage house speakers. Similarly, if your system is designed with larger speakers near the front of the house, having better low frequency output than the rear fill speakers, then those speakers should be fed the bass-heavy instruments. By adjusting the matrix mix level controls for the drum/percussion and bass guitar mixes so that more of these submixes goes to the matrix outputs that feed the near-stage speakers, and less to the rear fill speakers, the overall sound quality in the house will be improved.

For program fades, you have a choice: you can use the Mix Master Faders [35], in which case the previously established balance for each zone of the sound system reappears as soon as these Faders are returned to their correct settings. Or you can use the MATRIX MASTER controls [39], in which case the previously established program (mix) balance remains, but you'll have to recreate the zone-to-zone balance when you bring up the MATRIX MASTER controls. Of course, you can always use the Mix ON/off switches [34] or Matrix ON/off switches [41] to mute the output to the speaker system, thereby eliminating any uncertainty in re-establishing program levels.

If the PM2800M internal jumpers are arranged so that the Mix-to-Matrix feeds are derived pre-fader (as described in Section 6.4), then the Mix Master Faders [35] will not affect the matrix mix levels. In this case, the matrix can be used in much the same way, to create the necessary mono or stereo house feeds, while

the mix outputs can be adjusted independently to feed a multitrack tape recorder. (The stereo output is also useable here, though it may or may not have a useful program balance since the mix levels that feed it may be optimized for the matrix, not for a specific mix.) Whereas the signals applied to tape are generally recorded at a uniformly "hot" level (high enough to optimize signal-to-noise ratio, and just low enough to avoid saturation), the same mix signals can be mixed to achieve the desired program balance for the live sound presentation.

SECTION 9 MAINTENANCE

9.1 CLEANING THE CONSOLE

9.1.1 The Console and Power Supply Exterior

The console and power supply are painted with a durable finish. To avoid damage to the paint, control knobs, switch caps and other parts, **DO NOT USE SOLVENTS**. Instead, keep the console as free of dust as practical. Cover it when not in use, and brush or vacuum it periodically. The surface may be cleaned with a soft rag moistened with a dilute solution of non-abrasive detergent and water. If sticky gum is left on the panel (from masking tape or other tape used for channel labeling), it may be necessary to use a specialized solvent. In general, rubber cement solvent will remove tape residue without harming the console; however, it is your responsibility to test any such solvent in an inconspicuous location to ensure it does not attack the console finish or mar any plastic part.

Avoid getting the inside of the console wet from excessively wet rags. **DO NOT USE AEROSOL OR SPRAY CLEANERS**.

9.1.2 Power Supply Air Filter

The reticulated foam air filter on the front of the PW2800 power supply screens cooling air as it is drawn through the unit. When the foam becomes clogged or dirty, it should be cleaned; check it periodically. Using a 3 mm Allen wrench, remove the four cap screws that secure the front grille. The foam element may now be removed and rinsed in cool water. For greasy or stubborn dirt, dip the element in a mild solution of detergent and water, then rinse with clear water. Blot and/or air dry the element thoroughly before returning it to the power supply. **DO NOT USE SOLVENTS TO CLEAN THE FOAM ELEMENT**.

9.1.3 Pots and Faders

Yamaha **DOES NOT** recommend the routine use of any contact cleaners or solvents for cleaning pots or faders. Such "preventive maintenance" can actually do more harm than good by removing the lubricating film on certain pots or faders. While treatment with such solvents or cleaners may temporarily "clean up" a noisy control, it can also quickly result in a worn element (due to lack of lubrication) and even greater, incurable noise.

When a component is to be cleaned, use a very small amount of an appropriate cleaner, solvent, or

pure isopropyl alcohol. Try to get it on the element, and immediately work the pot or fader several times all the way between stops.

One safe practice that may reduce dirt and dust build up is to rapidly move each control on the console through its full range of travel. Don't "slam" the faders or rotary knobs against their stops, just move them all the way back and forth. In general, cleaning pots and faders is not a trivial task. Some have carbon elements, some have conductive plastic elements, and others have cermet elements. What cleans one part reliably may not work on another. When in doubt, consult your authorized Yamaha PM2800M dealer or service center.

9.1.4 The Console Interior

Dust and dirt are the enemy of electronic and mechanical systems. Switches and controls may wear prematurely due to the abrasive nature of dirt. A coating of dust may, in some cases, be conductive and change the electrical properties of the circuit. Similarly, dirt accumulations can reduce the thermal dissipation from heat sinks and transistors, leading to premature failure. It is advisable to use a soft brush or a vacuum cleaner with a soft brush attachment to clean the console periodically. Depending on the environment, this may be as often as once a month, or as infrequently as once a year. Use care not to bend or dislodge any components. Always do this work with the console power **OFF**.

If a beverage is spilled into the console, try to blot up as much excess moisture as possible immediately. If practical, immediately turn off the power and remove any affected modules. If not, wait until it is practical, and then turn off the power and proceed. Rinse contaminated parts on the module with distilled water, shake off the excess water, blot dry with a soft cloth, and air dry or use a warm (not hot) stream of air from a hair dryer to facilitate drying. If the console interior is contaminated, wipe it clean with a water-moistened cloth.

It is best to clean a spill as soon as possible. Unsweetened black coffee is probably the least harmful. The sugar in sweetened coffee can leave a sticky film on parts, and cream or milk will leave a residue that can be very troublesome. Similarly, sweetened soft drinks and fruit juices can leave sticky residues that degrade the performance of switches, faders and pots.

9.1.5 Module Removal and Replacement (see OPTIONAL FUNCTIONS, Section 6.)

9.2 METER LAMP REPLACEMENT

Two lamps illuminate the face of each VU meter.

To change a meter lamp, first open the meter bridge. This is done by removing 2 screws from the side and 4 screws from the rear panel behind the meter bridge. The bridge is hinged on the rear panel, and can be swung open for access to the meters.

Each replacement lamp (Yamaha part number 75570) comes with a connector affixed to pigtail leads from the lamp. Withdraw the old lamp from the rear, pulling it out of its retaining grommet in the meter face, and unplug the connector from the rear of the meter assembly. Insert the new lamp in its place, and secure the connector.

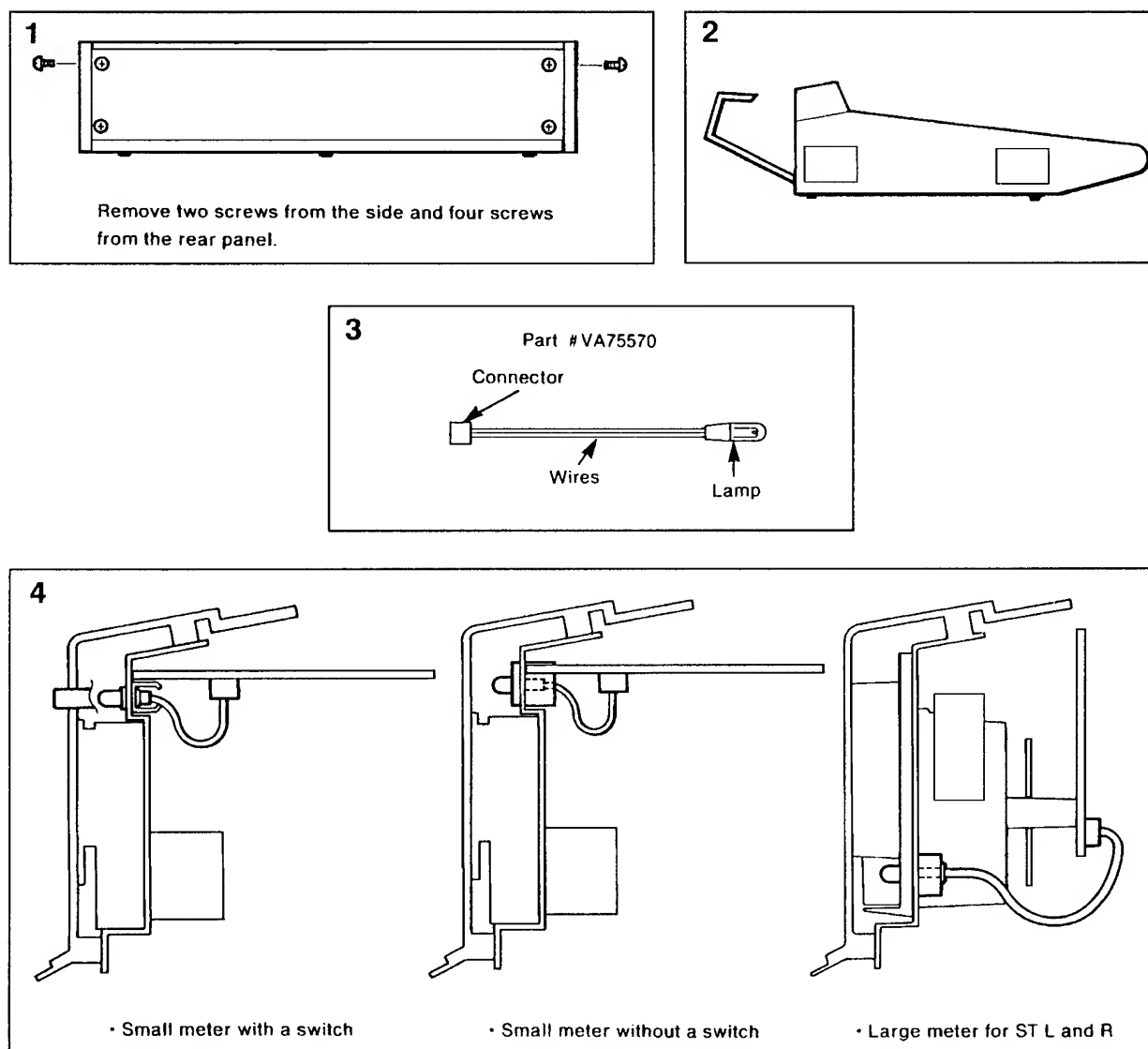


Figure 9-1. Replacement of VU Meter Lamps.

9.3 WHERE TO CHECK IF THERE IS NO OUTPUT

In general, when something appears not to be working properly in a sound system, it is necessary to have a clear understanding of the system block diagram. One should look for a "good" signal by patching around suspect equipment, modules or circuits. Suspected "bad" cables can be replaced or swapped to see if the problem follows the cable. These techniques should be known to most experienced

sound system operators. In the case of the PM2800M console, however, there are a number of apparent fault conditions, which the operator may inadvertently create simply by setting controls in a particular configuration, whereby no signal reaches the output. The following chart depicts the most likely errors you may encounter, and points out how to correct the problem.

"FAULT" CONDITION	POSSIBLE CAUSE	CORRECTION
Input channel signals do not appear at the Mix, Stereo, Aux or Matrix outputs.	Console is in SOLO mode, and an input channel to which no signal is applied has its CUE/SOLO switch engaged.	Release the master SOLO MODE switch to activate all channels which have CUE switches engaged.
	The affected input channel(s) have MUTE assign switches engaged, and the MASTER MUTE group to which the channel(s) is assigned is set to mute mode.	Disengage the MASTER MUTE switch or the affected input channel MUTE switch(es).
	The affected input channel(s) has one or more MUTE assign switches engaged, and the remote MUTE connection is causing the MASTER MUTE group to be engaged.	Disconnect the MUTE CONTROL connector to isolate the cause of the problem; if output is restored, check the remote circuitry.
Certain input channels or groups of channels cannot be heard at the Mix outputs, and do not feed the matrix via the MIX-TO-MATRIX or ST-TO-MATRIX switches.	The affected input channels are assigned to a Mix Fader (or Stereo Fader) which is set to minimum level (down), and the M>MTRX (or ST>MTRX) feeds are jumpered to be post-fader.	Raise the Mix Fader (or Stereo Fader) setting to a higher level.
Individual input channel cannot be heard at the Mix, Stereo, Aux or Matrix outputs.	Channel ON/off switch is off, or its PAD and GAIN controls are set so that input sensitivity is very low. A phantom powered condenser microphone or direct box is connected to the channel and is not receiving phantom power.	Turn ON the channel. Set the PAD to a lower value (less dB of attenuation – i.e., 0 instead of 20) and/or the GAIN to a higher value.
		Check to be sure the channel and master 48V switches are on.
There is no output signal, and no console functions work at all.	Power is not reaching the PM2800M.	Verify that the PW2800 is ON and that its umbilical cable is properly connected. Check the power supply fuse, and check the power outlet for presence of the correct AC mains voltage.

9.4 WHAT TO DO IN CASE OF TROUBLE

The PM2800M is supported by Yamaha's world-wide network of factory trained and qualified dealer service personnel. In the event of a problem, contact your nearest Yamaha PM2800M dealer. For the name of the nearest dealer, contact one of the Yamaha offices listed below.

Yamaha Corporation
10-1 Nakazawa-Cho
Box 1
Hamamatsu, 430 Japan

Yamaha Europa G.m.b.H.
2084 Rellingen b.
Hamburg, Siemensstr. 22/34
West Germany

Yamaha Musique France S.A.
Rue ambroise croizat,
parc d'activites de Paris Est,
77200 Torcy-marne la Vallee,
France

Yamaha-Kemble Music (U.K.) Ltd.
Mount Ave., Bletchley,
Milton Keynes MK1 1JE,
England

Yamaha Scandanavia AB
Fröfästegatan 1,
(Box 300 53)
400 43 Göteborg, Sweden

Yamaha Music Australia
17-33 Market Street
South Melbourne, 3205
Australia

Yamaha Canada Music Ltd.
135 Milner Avenue
Scarborough, Ontario
M1S 3R1, Canada

Yamaha Corporation of America
6600 Orangethorpe Avenue
Buena Park, CA 90620
U.S.A.

IMPORTANT NOTICE FOR THE UNITED KINGDOM


Connecting the Plug and Cord

WARNING : THIS APPARATUS MUST BE EARTHED

IMPORTANT. The wires in this mains lead are coloured in accordance with the following code:


GREEN-AND-YELLOW	:	EARTH
BLUE	:	NEUTRAL
BROWN	:	LIVE

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug proceed as follows:

The wire which is coloured GREEN-AND-YELLOW must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol  or coloured GREEN or GREEN-AND-YELLOW.

The wire which is coloured BLUE must be connected to the terminal which is marked with the letter N or coloured BLACK.

The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

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